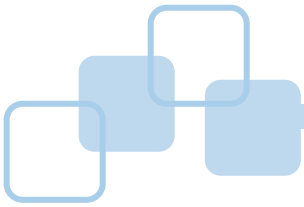

USER'S MANUAL

VoIP Gateway

Venus 29XX Series SIP

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About This Manual

This section guides users on how to use the manual effectively. The manual contains information needed to install, configure, and operate TAINET's VENUS 29xx Series VoIP SIP Gateway. The summary of this manual is as follows:

Chapter 1: **Overview**

Chapter 2: **Hardware Installation**

Chapter 3: **Configuration and Management**

Chapter 4: **Edit Configuration File**

Chapter 5: **Troubleshooting**

Appendix A: **Supplemental Telephone Features**

Appendix B: **Abbreviation**

Appendix C: **Code Determination Flow Chart**

Symbols Used in This Manual

3 types of symbols may be used throughout this manual. These symbols are used to advise the users when a special condition arises, such as a safety or operational hazard, or to present extra information to the users. These symbols are explained below:

Warning:

This symbol and associated text are used when death or injury to the user may result if operating instructions are not followed properly.

Caution:

This symbol and associated text are used when damages to the equipment or impact to the operation may result if operating instructions are not followed properly.

Note:

This symbol and associated text are used to provide the users with extra information that may be helpful when following the main instructions in this manual.

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Chapter 1. Overview

1.1 Product Overview

TAINET VENUS 29xx Series, a critical part of the Next Generation Network (NGN), functions as an interface gateway unit between the traditional Public Switched Telephone Network (PSTN) and the IP packet network by performing the conversion of the analog voice traffic to/from the IP-based media stream.

VENUS 29xx Series is a Voice-over-IP (VoIP) Gateway that provides cost-effective and quality voice service over global IP packet network (Internet or Intranet) for end-users, and, in the meanwhile, lower the installation and maintenance effort. With analog voice interface connecting to user's existing telephone handset or PABX, and the Ethernet interface connecting to the service providers' IP network, VENUS 29xx Series bridges the PSTN and IP network.

Depending on the installed firmware, VENUS 29xx Series supports Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP) and MEGACO which comply with IETF RFC 3261, RFC 3435 and RFC 3525 respectively. This document is only for VENUS 29xx Series installed with SIP firmware.

VENUS 29xx Series allow traditional PSTN users to make phone calls over the Internet without changing their telephone operating procedure. Using a Proxy Server or SoftSwitch together with the VENUS 29xx Series, service providers or telecom companies are able to provide integrated voice and data service over the IP network to their customers. Features like billing, accounting and network management supported by the SoftSwitch are available as well.

VENUS 29xx Series installed with SIP firmware provides the following services and functions:

- Support SIP protocol, which complies with IETF RFC 3261 standard.

- Venus 2908/2916/2924/2932 Series support up to 8/16/24/32 analog voice interfaces respectively for traditional POTS (Plain Old Telephone System) equipments (telephone handset, FAX machines, etc).
- Venus 2908 Series provides 4 FXS/4 FXO ports or 8 FXS, Venus 2916/2924/2932 Series provides 16/24/32 FXS telephone interfaces and Venus 2932 FXO provides 32 FXO interfaces respectively for various customers. Each telephone interface can drive at least three telephone handsets through up to 4 km (13k feet) of 24 AWG copper lines.
- Support ITU-T standard voice codecs.
- Apply quality VoIP technologies including: Real-time voice packet encapsulation and voice play-out based on the Real-Time Protocol (RTP), Adaptive Jitter Buffer, Echo Cancellation（ITU-T G.168/165）, Compensation for Loss of Packet, Voice Activity Detection (VAD) and Silence Suppression.
- Support T.30 Transparent FAX service or T.38 FAX Relay.
- Supplemental Class 5 features like Call Forwarding, Caller ID Display/Blocking, Call Returning, Speed Dial, Call Waiting, Call Transfer, Voice Message Indication, Hotline, Call Park and 3-way Conferencing… etc can be easily supported.
- Use static IP address assignment or obtain the dynamic IP address from a DHCP (Dynamic Host Configuration Protocol) Server or BootP (Bootstrap Protocol) Server.
- Provide DHCP server function on LAN port.
- Provide PSTN lifeline feature as backup. (2908 only)
- Support PPPoE function (Point-to-Point Protocol over Ethernet), NAT/NAPT function (Network Address Translation/Network Address Port Translation), port-based VLAN, VLAN ID and priority tagging, and QoS function that including IEEE 802.1p and IEEE 802.1Q
- Support RFC-2833 RTP payload for DTMF Digits, Telephony Tones and Telephony Signals
- Provide Terminal User Interface via the console port、TELNET or web browser for easy and quick local or remote configuration and monitoring.
- Provide SNMP MIB for integrating into service providers' SNMP management system.

- Support TFTP protocol for remote software upgrade and configuration file download.
- User Interface is protected with User ID and Password from unauthorized users.
- Provide three user accounts with three levels of access privilege.
- Provides a flexible dialing principles, as well as speed dial using hot key, and hotline direct dial functions.
- Support SRTP, Secure Real-time Transport Protocol, for enhanced security.
- Support TR-069 Management Protocol.
- RoHS compliant products.

1.2 Applications

VENUS 29xx Series, designed as a desktop or rack-mount unit, can be easily installed in a telecommunication equipment room of a building. It uses the standard 10/100 Base-T Ethernet to connect to the service providers' IP backbone through the router or data communication equipment. Up to 32 traditional telephone handsets or FAX machines located in the rooms at each floor of the building can be connected to the VENUS 29xx Series telephone ports using copper wires (Figure 1-1). With various management features supported on VENUS 29xx Series, service providers or telecomm companies can provide toll-quality voice service to the customers. The deployment of an intelligent IP packet network for integrated data and voice service with billing, accounting, and management features can be soon built up.

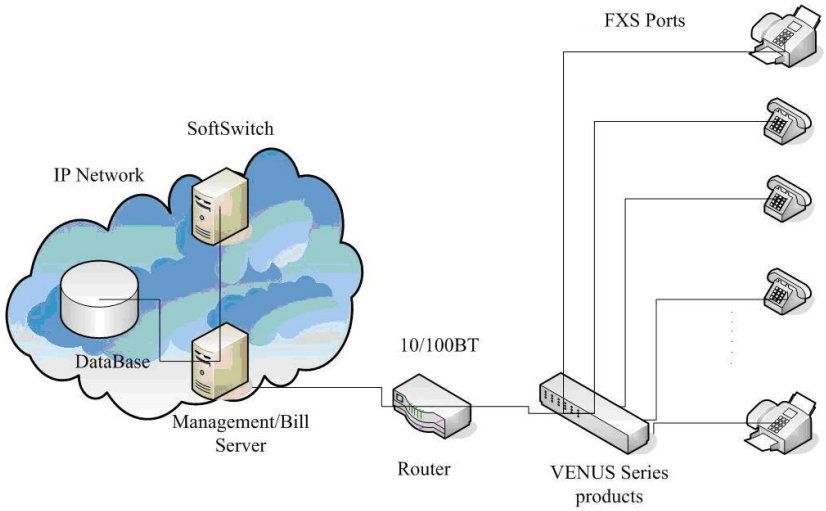


Figure 1-1 Typical SIP VoIP Application

Large or median sized enterprises are able to build-up the high quality, cheap or free, private voice service based on the Intranet or Extranet network by deploying VENUS 29xx Series units at corporate HQ and local branch offices. The VENUS 29xx Series communicates with the SoftSwitch/SIP Proxy Server installed at corporate HQ through managed IP leased line, such as DDN (Digital Data Network) leased line or VPN (Virtual Private Network) network. With the scalable voice capacity feature provided by VENUS 29xx Series, enterprises can install proper amount of telephone interfaces for each branch (Figure 1-2).

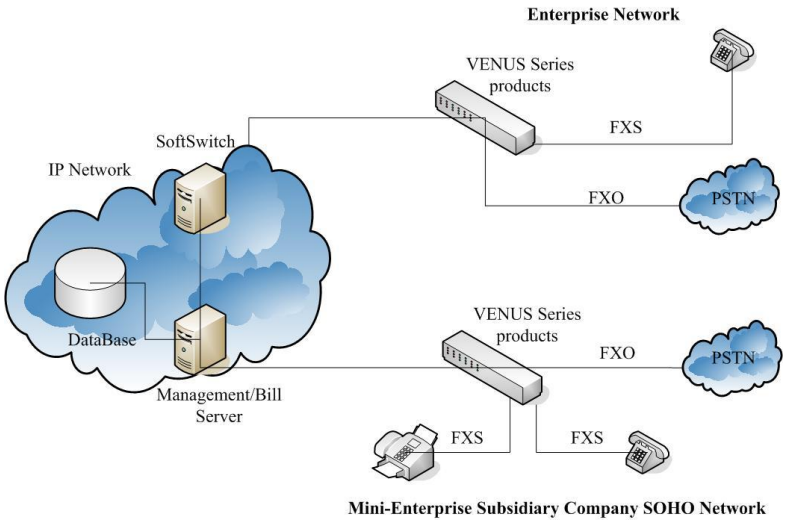


Figure 1-2 Enterprise VoIP Intranet Application

For the Multi-Tenants Units (MTU) or Multi-Dwelling Units (MDU) markets, such as a hotel or campus, VENUS 29xx Series performs as a voice access platform to accommodate the voice-over-IP traffics from all the users. Thanks to the convenience of adding new Ethernet nodes, new VENUS 29xx Series units can be installed easily to extend the amount of users. To support more VENUS 29xx Series media gateway allows service providers to provide cost effective and quality voice services with minimum capital investment and maintenance cost.

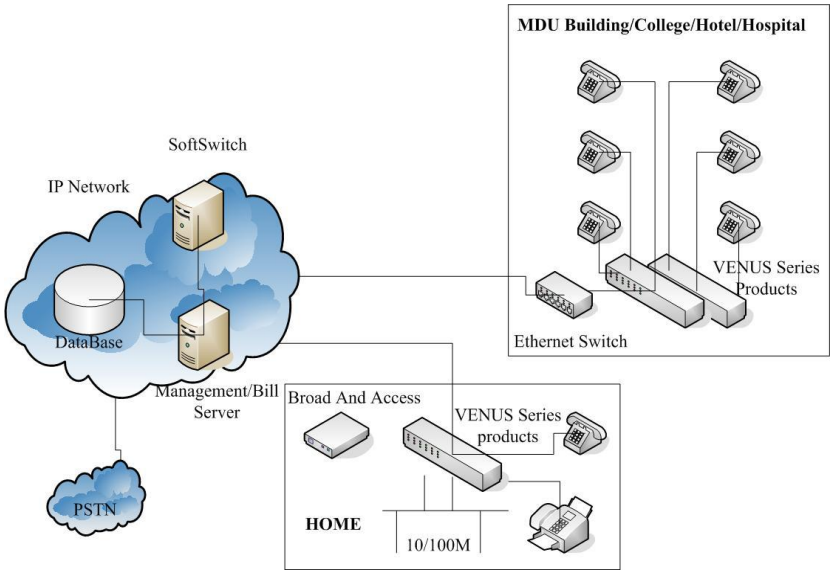


Figure 1-3 MTU/MDU Application

The advantages of using VENUS 29xx Series, the high port density CPE (Customer Premise Equipment) Gateway, are

- Lower installation effort
- Easier maintenance

- Competitive per port price
- High voice quality
- Available feature extension
- Consuming less public IP addresses which are valuable resources

1.3 Product Description

1.3.1 Hardware Architecture

VENUS 2932 is a compact standalone VoIP Gateway with AC power, built-in 32 channels Voice Processing capacity including 32 FXS ports or 32 FXO ports.

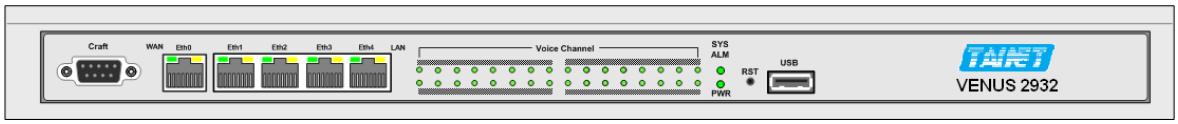


Figure 1-4 Front View of VENUS 2932

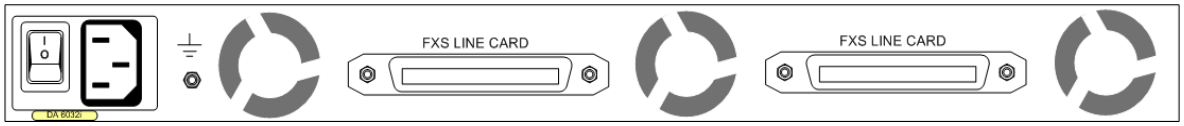


Figure 1-5 Rear View of VENUS 2932

VENUS 2924 is a compact standalone VoIP Gateway with AC power, built-in 24 channels Voice Processing capacity.

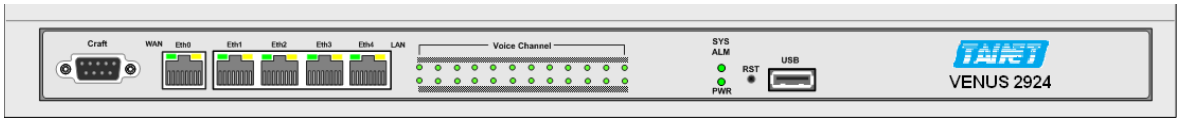


Figure 1-6 Front View of VENUS 2924

Figure 1-7 Rear View of VENUS 2924

VENUS 2916 is a compact standalone VoIP Gateway with AC power, built-in 16

channels Voice Processing capacity.

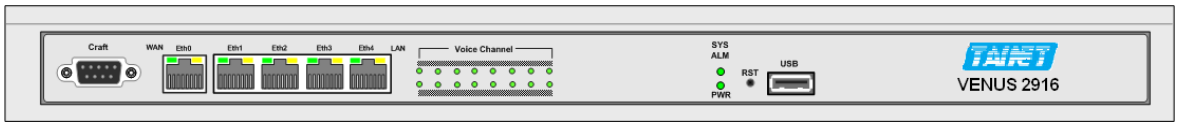


Figure 1-8 Front View of VENUS 2916

Figure 1-9 Rear View of VENUS 2916

VENUS 2908-8S is a compact standalone VoIP Gateway with AC power; built-in 8 channels Voice Processing capacity and 1 PSTN port for dial-line backup.

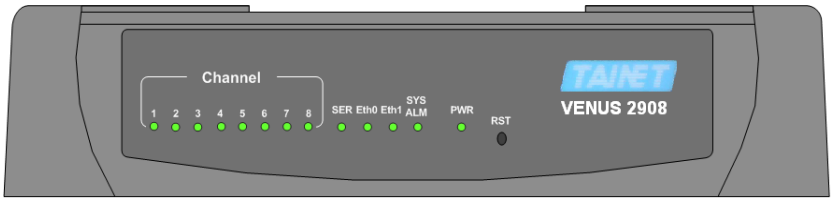


Figure 1-10 Front View of VENUS 2908-8S

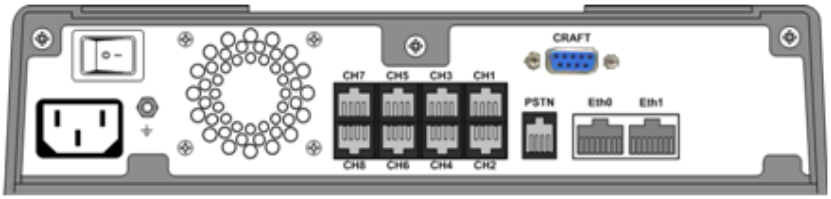


Figure 1-11 Rear View of VENUS 2908-8S

VENUS 2908-4S4O is a compact standalone VoIP Gateway with AC power; built-in 8 channels Voice Processing capacity including 4 FXS ports and 4 FXO ports.

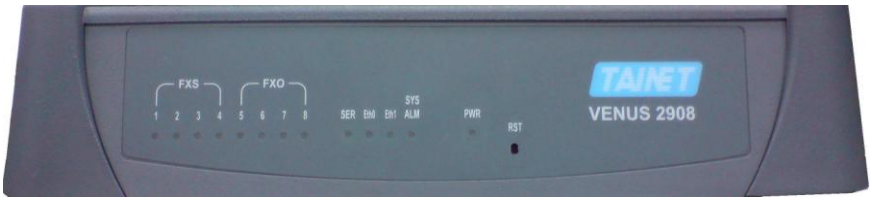


Figure 1-12 Front View of VENUS 2908-4S4O

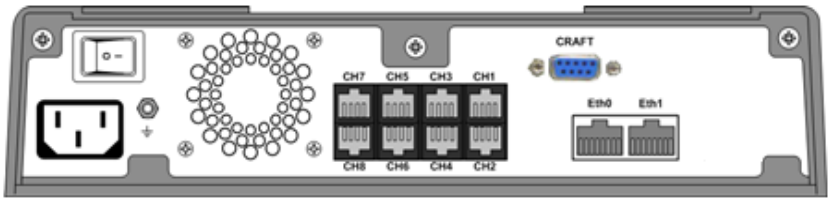


Figure 1-13 Rear View of VENUS 2908-4S4O

1.3.1.1 Venus 29xx Series Base Unit

- VENUS 2908：Base Unit with **AC** power supply
- VENUS 2916/2924/2932：Base Unit with optional **AC or DC** power supply

1.3.1.2 Accessories

- 50-pin cable（Telco-Cable：RJ-21 male connector to 16 pairs of twisted pair converter for Venus2916/2932）
- 50-pin cable（Telco-Cable：RJ-21 male connector to 24 pairs of twisted pair converter for Venus2924）

1.3.2 Software Architecture

Applying advanced VoIP technology; VENUS 29xx Series uses powerful DSP (Digital Signal Processor) and CPU (Central Processing Unit) to build up a high performance Media Gateway platform. Running over an embedded Multi-tasking Real-Time Operating System, network protocols can be handled efficiently. Architecture of multiple DSPs for performing voice processing concurrently guarantees highest voice quality. Both the DSP and CPU software can be remotely upgraded for feature enhancement.

VENUS 29xx Series provides several different ways for equipment management:

- Terminal User Interface via the Craft port for local management
- Terminal User Interface via Telnet for remote management
- Terminal User Interface via Web browser fro remote management
- SNMP Management

- TR-069(CWMP) Management

1.3.2.1 Craft Port

By using the VT-100/ANSI compatible terminal emulation software, such as Microsoft HyperTerminal, user is able to configure VENUS 29xx Series via the Craft port at the rear panel.

1.3.2.2 Telnet

VENUS 29xx Series can be managed through a Telnet connection. The User Interface format and the management functions provided by TELNET are exactly the same as the craft port. To maintain the consistency of configuration, only one single user is allowed to login the terminal user interface via the craft port or Telnet at the same time.

1.3.2.3 Web Browser

The gateway allows users to make settings using a web browser, must enable Web function of the Venus configuration item first. After opening a browser, enter gateway's IP address as the website address in order to enter the Web configuration screen as shown in the following diagram. (IE Browser used for example)

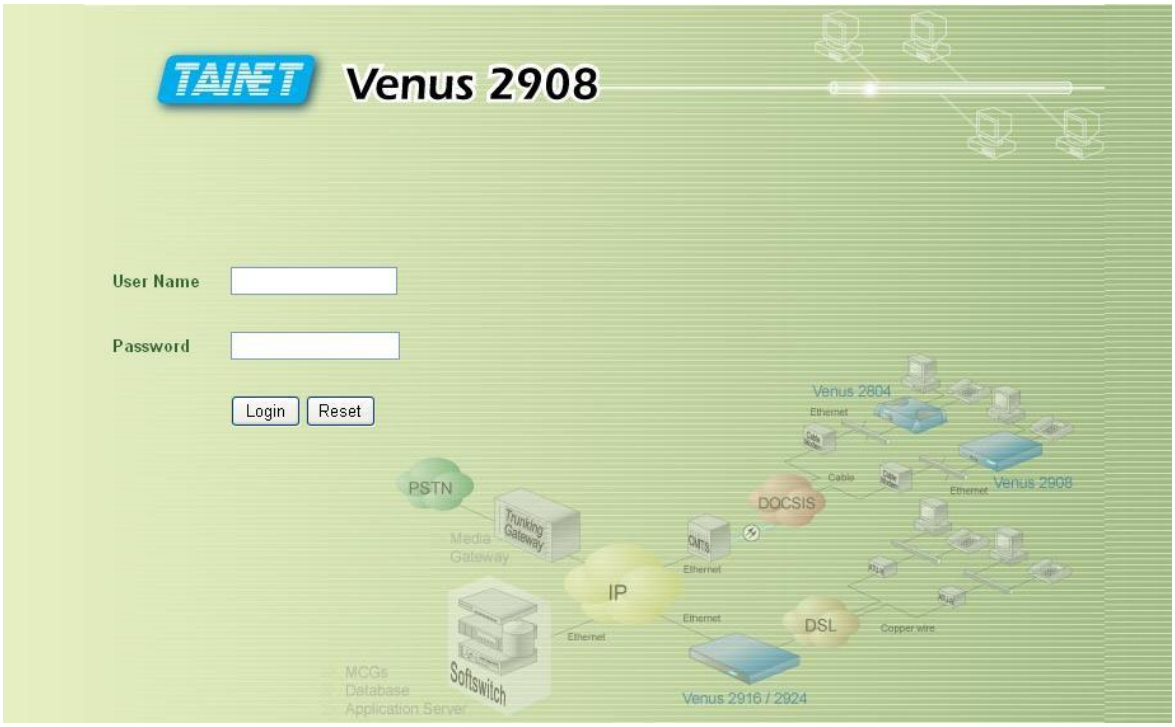


Figure 1-14 Web Browser

1.3.2.4 SNMP Management

The embedded SNMP agent in VENUS 29xx Series allows the device to be managed by the SNMP management system.

1.3.2.5 TR-069 Management

VENUS 29xx Series can be managed by the Auto Configuration Servers (ACS) via TR-069 protocol.

1.4 Technical Specifications

Table 1-1 VENUS 29XX Series Technical Specification

Items	Specification
Base Unit	• Number of slots for Ethernet switch module per unit: VENUS 2916/2924/2932(FXS/FXO): Built-in 5 Ethernet ports. VENUS 2908: Built-in 2 Ethernet ports.
	• Maximum Number of VoIP Channels supported: VENUS 2932(FXS/FXO): 32 FXS voice channels or 32 FXO voice channels. VENUS 2924: 24 voice channels. VENUS 2916: 16 voice channels. VENUS 2908-8S: 8 voice channels, 1 PSTN backup line VENUS 2908-4S4O: 4 FXS voice channels and 4 FXO voice channels.
	• Console Port: - Type: Ethernet Port, RS-232, DCE mode - Connector: DB-9 Female - Format: 115,200bps, no parity, 8 data bits, 1 stop bit
	• WAN Interfaces: IEEE 802.3u 10/100BaseT Fast Ethernet port with auto-negotiation
	• Indicators: - PWR LED: Power Status - SYS ALM LED: System Status and Diagnosis Indication - ETH(1~4) LED: LAN Network Card interface status - ETH0 LED: WAN Network Card interface status - Ser LED: Service status and PSTN status - VOICE CHANNEL 1~32 LED: Hook and Ring status of the voice channels
	• RST Push Button: RESET
	• Power Supply - AC power model: 100V ~ 230VAC, 50~60 Hz - DC power model: -36V ~ -72VDC
FXS Telephone Interface channel (FXS-8, FXS-4 and FXS-32)	• Signaling Type: FXS loop start driving 2-wire analog telephone handsets or G3 FAX machines
	• Input Impedance: 600-ohm termination mode
	• Return Loss: > 20dB from 200Hz to 3.4kHz
	• Attenuation/Frequency Distortion: ITU-T G.712 Compliant

Items	Specification
	• Group Delay: ITU-T G.712 Compliant
	• Total Distortion: ITU-T G.712 Compliant
	• Longitudinal balance: <45 dB from 200Hz to 3.4kHz
	• Gain: - A/D: -3 +- 0.2dB at 1KHz - D/A: -3 +- 0.2dB at 1KHz
	• DC Feed Current: 25mA current limited
	• Line Current Detection: <10mA
	• Ringer Frequency: 20Hz (17~40Hz selectable)
	• Ringer Output Voltage: >40Vrms, 3 REN
	• Ringing Current Limit: 100mA current limited
	• Signaling Format: DTMF or pulse dial
	• Idle State Voltage: Vtip-ring < 48V, and Vring<Vtip < 0V
	• 50-pin female RJ-21 Telco connector
	• Optional fan-out cable (50-pin connector to 16 or 24 RJ-11 plugs) for wiring or testing
	• Long haul (up to 3,000m of 24 AWG lines) design
FXO Telephone Interface channel (FXO-4 and FXO-32)	Electronic Spec
	• Signaling Type: FXO Loop Start
	• Line Impedance: 600-ohm termination mode
	• Return Loss: > 20dB from 200Hz to 3.4KHz
	• Attenuation/Frequency Distortion: ITU-T G.712 Compliant
	• Group Delay: ITU-T G.712 Compliant
	• Total Distortion: ITU-T G.712 Compliant
	• Longitudinal balance: <45 dB from 200Hz to 3.4kHz
	• Gain: - A/D: -3 +- 0.2dB at 1kHz - D/A: -3 +- 0.2dB at 1kHz
	• Leakage Current < 10uA
	• On-hook Resistance > 10Mohm
	• DC Current Limit: 133mA
	• Caller ID Detection
	• Signaling Format: DTMF
	• 50-pin female RJ-21 Telco connector
	Ring Detection
	• Ring Detection Frequency: 17~40Hz (configurable)

Items	Specification
	• Ring De-bounce: < 200ms (configurable)
	• Ring Cadence: - Inter-Pulse < 550ms (Configurable) - Inter-Cycle < 5000ms (Configurable)
	• Auto Answer after Ring: 1 ring (Configurable)
	Line Disconnection
	• Detection of the Disconnection Tones: (User Configurable) - Busy Tone - Reorder Tone - Congestion Tone
	• Long Dial Tone without Dialing: 10 seconds
	• Line Reversal Detection
Voice Codec	• ITU-T G.711 μ-Law (64kbps)
	• ITU-T G.711 A-Law (64kbps)
	• ITU-T G.723.1 (6.3K/5.3kbps)
	• ITU-T G.729A (8kbps)
	• ITU-T G.726 (16k/24k/32k/40kbps)
Tone Generation and Detection	• DTMF, Call Progress tones (Dial tone, Busy tone, Ring back tone, Congestion tone, Off-hook notice tone, etc)
	• V.21/V.25/V.8 tone detection for automatic data/fax/voice switch over to G.711 voice coder
Quality Enhancement Technologies	• Voice Activity Detection (VAD)
	• Silence Suppression
	• Comfort Noise Generation (CNG)
	• Adaptive Jitter Buffer
	• Echo Cancellation (ITU-T G.165/G.168 compliant)
	• Compensation for Loss of Packet
	• In-band or Out-band DTMF Relay
	• Selectable TX/RX Gain Controls
Call Signaling (Call Control) Protocols	• IETF RFC-3261 Session Initiation Protocol (SIP)
Voice Packet Encapsulation	• IETF RFC-1889 Real Time Protocol (RTP)
	• IETF RFC-2833 RTP payload for DTMF Digits, Telephony Tones and Telephony Signals
Media Description Protocol	• IETF RFC-2327 Session Description Protocol (SDP)

Items	Specification
Internet Protocols	• TCP/IP, UDP, ARP/RARP, ICMP, Telnet, DNS Client, BootP Client, DHCP Client/Server, SNMP Client,CWMP Client, TFTP Client, PPPoE, RIP I/II and NAT
QoS	• IEEE 802.1p
VLAN	• VLAN (ID, priority) tagging, port-based VLAN
IP Address Assignment	• Static or Dynamic using DHCP or BootP client
FAX over IP	• Fax Auto-detection • T.30 Transparent FAX • FAX Relay Mode: ITU-T T.38 Real Time G3 FAX over IP protocol supporting ITU-T V.21, V.27ter, V.29 and V.17 up to 14,400bps
Management	• Console, Telnet, Web Browser, proprietary authentication，SNMP management，TR069 management
Remote Upgrade & Remote Provisioning	• Software download and Configuration file download/upload via TFTP
Circumstance	• Operating Temperature: 0°C ~ 50°C • Storage Temperature: -10°C ~ 70°C • Relative Humidity: 10% ~ 90% (non-condensing)
Physical Dimension	• 437 W x 44 H x 286 D mm (2916/2924/2932) • 250 W x 55 H x 200 D mm H (2908)
EMC Compliant	• EN 50081-1, EN 50081-2 • FCC Part 15 Class A • CE mark
Safety Compliant	• EN-60950 • FCC Part 68 • UL/CSA

1.5 Standard Compliance

VENUS 29xx Series complies with the following standards and protocols:

- ITU-T Codec Standard
- ITU-T V.21/V.25/V8 – FAX / Modem Answer Tone
- ITU-T G.168/G.165 Digital network echo cancellers
- IETF RFC 3261 – SIP (Session Initiation Protocol)
- IETF RFC 3262 – Reliability of Provisional Responses in Session Initiation Protocol (SIP)
- IETF RFC-3263 – Locating SIP Servers
- IETF RFC 3264 – An Offer/Answer Model with Session Description Protocol (SDP)
- IETF RFC 3265 – SIP - Specific Event Notification
- IETF RFC 1889 – RTP (Real Time Protocol)
- IETF RFC-2327 – Session Description Protocol
- IETF RFC-2543 – Session Initiation Protocol
- IETF RFC 2833 – payload for DTMF Digits, Telephony Tones and Telephony Signals
- IETF RFC 2976 – SIP INFO Method
- IETF RFC 3311 – SIP UPDATE Method
- IETF RFC-3323 – A Privacy Mechanism for the SIP
- IETF RFC-3325 – Private Extension to the SIP for Asserted identify within Trusted Networks
- IETF RFC 3515 – The SIP Refer Method
- IETF RFC-3550 – RTP-A Transport Protocol for Real-Time Applications. July 2003
- IETF RFC-3665 – SIP Basic Call Flow Example

- IETF RFC-3842 – A Message Summary and Message Waiting Indication Event Package for the SIP
- IETF RFC-3891 – The SIP "Replaces" Header
- IETF RFC-3892 – The SIP Referred-By Mechanism
- IETF RFC-3960 – Early Media and Ringing Tone Generation in the SIP
- IETF RFC-3986 – Uniform Resource Identifier: Generic Syntax
- IETF RFC-4028 – Session Time in the SIP
- IETF RFC-4566 – Session Description Protocol
- IP Protocols: TCP/IP, UDP, ARP/RARP, ICMP, Telnet, DNS Client, BootP Client, DHCP Client, DHCP Server, TFTP Client, SNMP agent (V1 or V2c),CWMP agent,,PPPoE, NAT/NAPT and RIP I/II

Chapter 2. Hardware Installation

2.1 Unpacking

This chapter provides the information for installation of the VENUS 29xx Series. Before unpacking, make a preliminary inspection of the container. Evidence of damage should be noted and reported immediately. Unpack the equipment as follows:

- Place the container on a flat surface and open the container.
- Carefully take the VENUS 29xx Series out of the container and place it securely on a flat, clean surface.
- Inspect the unit for signs of damage. Immediately report any damage found.
- Check the packing list against your order to ensure that the supplied modules match your order. If modules have been pre-installed in accordance with your order, check that all the modules are in their proper slots and are secure. Immediately report any deviations.
- Check that all the necessary items have been included, following items are shipped with your VENUS 29xx Series:
  - One VENUS 29xx Series Manual (this one)
  - One AC power cable

2.2 Site Requirements

2.2.1 Site Selection

Install the device in a clean area that is free from environmental extremes. Allow at least 6 inch (15.24 cm) in front of the device for access to the front panel, and at least 4-inch (10.2 cm) in back for cable clearance. Position the device so you can easily see the front panel.

2.2.2 AC Electrical Outlet Connection

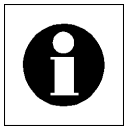
VENUS 29xx Series with AC power input should be installed within 1.83m (6 feet) of an easily accessible grounded AC outlet capable of furnishing the required supply voltage, in the range of 100 to 230V AC.

2.2.3 Grounding

The FCC requires telecommunications equipment to withstand electrical surges that may result from lightning strikes; the VENUS 29xx Series device meet the requirements set forth by the FCC. The following procedure outlines some common practices that can minimize the risk of damage to computer equipment from electrical surges.

- Make sure the electric service in your building is properly grounded as described in article 250 of the National Electrical Code (NEC) handbook.
- Verify that a good copper wire of the appropriate gauge, as described in Tables 250-94/95 of the NEC Handbook, is permanently connected between the electric service panel in the building and a proper grounding device such as:
  - A ground rod buried outside the building at least 8 feet (2.44 meters) deep in the earth.

- Several ground rods, connected together, buried outside the building at least 8 feet (2.44 meters) deep in the earth.
- A wire (see tables 250-94/95 of the NEC handbook for gauge) that surrounds the outside of the building and is buried at least 2.5 feet (.76 meters) deep in the earth.



Note:

The three grounding devices described above should be firmly placed in the earth. Soil conditions should not be dry where the device is buried.

- If you are unsure whether the electric service in your building is properly grounded, have it examined by your municipal electrical inspector.
- Install a surge protector between the device and Ground point. Any additional computer equipment you have connected to the device (directly or through another device), such as a terminal or printer should also be plugged into the same surge protector. Make sure that the surge protector is properly rated for the devices you have connected to it.
- Call your telephone company and ask them if your telephone line is equipped with a circuit surge protector.
- If you are operating the device in an area where the risk of electrical surges form lightning is high, disconnect the device from the telephone line at the rear panel when it is not in use.

2.3 LED Indicators

Table 2-1 LED Description

Label	Function Description	Colors
PWR	Power Status	Off (Power is Off)
		Green (Power is ON)
SYS ALM	System Status	Green (Normal Operation)
		Yellow (Performing Diagnosis)
		Red (System Failure)
VOICE CHANNEL LEDs	Voice Channel Status	Off (On-Hook State of the Telephone Handset)
		Green (Off-Hook State of the Telephone Handset)
Eth0	Ethernet 0 Status	Green (Network Card is plug in and 10/100 Base-T Link is Up)
		Off (Network Card is unplug)
Eth1(2908) Eth1~4(2916/ 2924/2932)	Ethernet (1~4) Status	Green (Network Card is plug in and 10/100 Base-T Link is Up)
		Off (Network Card is unplug)
SER(2908)	Server Status	Green (Registered to the SIP server)
		Off (Not registered to the SIP server)

2.4 Cable Connection

2.4.1 Connecting the Power Source

VENUS 29xx accepts 100V~230VAC/50~60Hz (Auto-range) power source. Connect the device to a power outlet of the correct voltage and rating as your power system.

Note:

It is strongly recommended to use an AC power cord with Grounding pin or connect the Grounding Screw on the rear panel to the grounded supply of correct power system.

2.4.2 Connecting the IP Network via Ethernet

On the base unit of VENUS 29xx Series, the embedded 10/100Base-T Ethernet port is provided as the standard interface to the IP network. The pin layout of the RJ-45 connector for IEEE 802.3 standard 10/100Base-T Ethernet ports are defined as following:

Table 2-2 10/100Base-T Connection

Pin #.	Pin Function	Pin #.	Pin Function
1	TD+	5	N/C
2	TD-	6	RD-
3	RD+	7	N/C
4	N/C	8	N/C

For connecting the 10/100Base-T Fast Ethernet, a Category 5 unshielded twisted-pair (UTP) cable or shielded twisted-pair cable is used. Two pairs of the twisted wires are used for separated Rx (reception) and Tx (transmission). The Fast Ethernet port is backward compatible with traditional 10Base-T Ethernet. VENUS 29xx Series can automatically detect whether it is connected to a 10Base-T or 100Base-T Network.

2.4.3 Connecting the Terminal

The Console port connector labeled “CRAFT” on the front panel of VENUS 29xx is provided for connection to an external ANSI or VT-100 compatible terminal for quick and easy, local configuration of the VENUS 29xx Series.

Speed and Data format: 115,200bps, none parity, 8 data bits, 1 stop bit, and no flow control.

2.4.4 Connecting the Telephony Devices

VENUS 2908 supplies 8(2908-8S) or 4(2908-4S4O) telephony ports.

Figure 2-1 VENUS 2908-8S Series Telephony Interface

Figure 2-2 VENUS 2908-4S4O Series Telephony Interface

Venus 2916 used the first RJ-21 interface, with the pin configuration as shown in the table below.

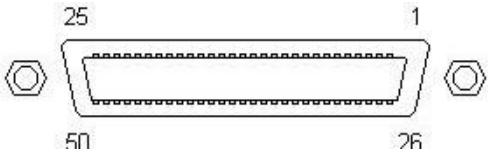


Figure 2-1 RJ-21 connector

Table 2-3 Venus2916 RJ-21Pin Assignment

Pin #	Port
1， 26	Port 1
2， 27	Port 2
3， 28	Port 3
4， 29	Port 4
5， 30	Port 5
6， 31	Port 6
7， 32	Port 7
8， 33	Port 8
9， 34	Port 9
10， 35	Port 10
11， 36	Port 11
12， 37	Port 12
13， 38	Port 13
14， 39	Port 14
15， 40	Port 15
16， 41	Port 16
17 ~ 25	Reserve
42 ~ 50	Reserve

Venus 2924/2932 used the first RJ-21 interface, with the pin configuration as shown in the table below.

Table 2-4 Venus2924/2932 First RJ-21Pin Assignment

Pin #	Port
1， 26	Port 1
2， 27	Port 2
3， 28	Port 3
4， 29	Port 4
5， 30	Port 5
6， 31	Port 6
7， 32	Port 7
8， 33	Port 8
9， 34	Port 9
10， 35	Port 10
11， 36	Port 11
12， 37	Port 12
13， 38	Port 13
14， 39	Port 14
15， 40	Port 15
16， 41	Port 16

17， 42	Port 17
18， 43	Port 18
19， 44	Port 19
20， 45	Port 20
21， 46	Port 21
22， 47	Port 22
23 48	Port 23
24， 49	Port 24
25， 50	Reserved

Venus 2932 used the second RJ-21 interface, with the pin configuration as shown in the table below.

Table 2-5 Venus2932 Second RJ-21Pin Assignment

Pin #	Port
1， 26	Port 17
2， 27	Port 18
3， 28	Port 19
4， 29	Port 20
5， 30	Port 21
6， 31	Port 22
7， 32	Port 23
8， 33	Port 24
9， 34	Port 25
10， 35	Port 26
11， 36	Port 27
12， 37	Port 28
13， 38	Port 29
14， 39	Port 30
15， 40	Port 31
16， 41	Port 32
17 ~ 25	Reserve
42 ~ 50	Reserve

There are two cables available:

- Twisted-pair cable: Male RJ-21 connector to 16 / 24 pair twisted pair cable.
- Fan-out cable: RJ-21 connector to 16 / 24 RJ-11 copper cable.

Twisted-pair cable: Male RJ-21 to 16 /24 pair twisted pair cable can easily be connected to the line patch panel.

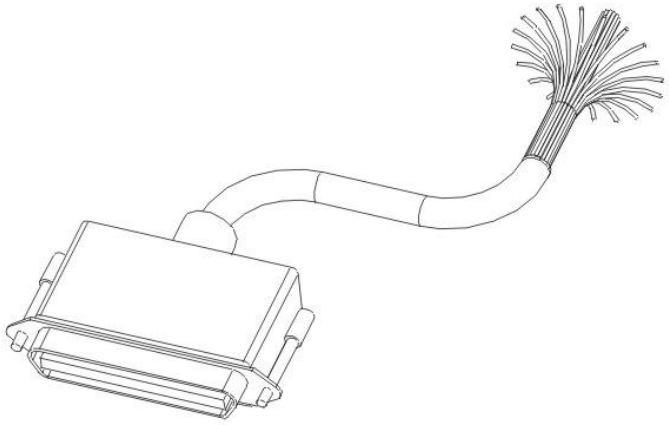


Figure 2-3 Twisted Pair Cable

Table 2-6 Twisted Pair Pin Assignment

RJ-21 Pin #	Copper Color	Voice Channel（per pair）
1 & 26	Black / Black and White	Port 1
2 & 27	Brown / Brown and White	Port 2
3 & 28	Red / Red and White	Port 3
4 & 29	Orange / Orange and White	Port 4
5 & 30	Yellow / Yellow and Black	Port 5
6 & 31	Green / Green and White	Port 6
7 & 32	Blue / Blue and White	Port 7
8 & 33	Purple / Purple and White	Port 8
9 & 34	Gray / Gray and Black	Port 9
10 & 35	White / Orange and Black	Port 10
11 & 36	Pink / Red and Black	Port 11
12 & 37	Light Green / Green Black	Port 12
13 & 38	Pink and Black / Pink and Red	Port 13
14 & 39	Pink and Blue / Pink and Green	Port 14
15 & 40	Light Blue / Light Yellow	Port 15
16 & 41	Light Blue and Black / Light Blue and Red	Port 16
17 & 42	Light Blue and Blue / Light Blue and Green	Port 17
18 & 43	Gray and Red / Gray and Green	Port 18

RJ-21 Pin #	Copper Color	Voice Channel （per pair）
19 & 44	Blue and Black / Purple and Black	Port 19
20 & 45	Light Green and Black / Light Green and Red	Port 20
21 & 46	Light Green and Blue / Light Green and Green	Port 21
22 & 47	Light Yellow and Black / Light Yellow and Red	Port 22
23 & 48	Light Yellow and Blue / Light Yellow and Green	Port 23
24 & 49	White and Black / White and Red	Port 24
25 & 50	White and Blue / White and Green	Reserved

2.5 Hardware Diagnosis

When the connections are all done, turn on the power of the device to boot up the software. VENUS 29xx Series will perform hardware detection, initialization and diagnosis. The LED indicators at the front panel show the progress of the hardware diagnosis. If it passed all the hardware diagnosis, the “SYS ALM” LED will display GREEN color; whereas, the RED color indicates hardware failure. The diagnosis result can be observed via the User Interface (UI) provided by VENUS 29xx Series.

Chapter 3. Configuration and Management

This Chapter describes the Terminal User Interface provided by VENUS 29xx Series. There are three methods to access to the Terminal User Interface: the Craft port, Telnet and Web browser, those present the exactly same format of terminal management. The Craft port is used primarily when the device is installed for the first time and the IP configuration is not yet provisioned. Once when the IP connection is provisioned, user may login to the Terminal User Interface using the Telnet software or Web browser to remotely control or maintain the device from anywhere in the global IP network.

3.1 Quick Start

Once when the VENUS 29xx Series is successfully started up, please make sure **that the connection to the IP network is properly installed** before continuing with the configuration. Very few parameters are required for configuring the VENUS 29xx to bring up basic VoIP service. Here is a guide for start VoIP service quickly.

3.1.1 Connecting to Venus 29xx

3.1.1.1 Setup PC

There are five ways to connect the VENUS 29xx, the Web-based management, Telnet_based management, SNMP_based management, TR069_based management and the Terminal Emulation-based management. The Web-based one is recommended.

The connection between PC and VENUS 29xx SERIES as shown in the following Figure:

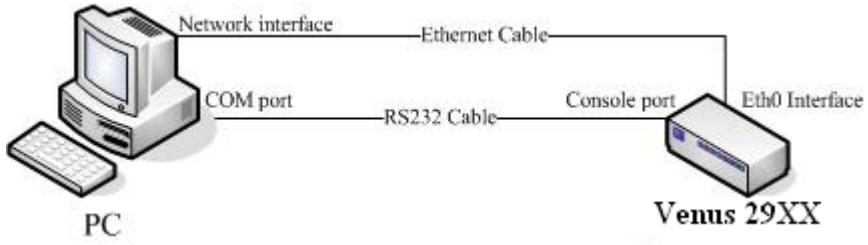


Figure 3-1 Connecting to the PC

1) Web-based

The web-based management allows users to manage VENUS 29xx from a remote Web browser. The embedded web server in the VENUS 29xx Series provides a user-friendly interface.

You can complete the configuration by web. First, you need to connect VENUS 29xx Series Eth0 interface to the network interface of your PC with Ethernet cable. Then, you need web browser software, such as Internet Explorer (IE), Mozilla Firefox, Netscape Navigator, etc to enter the web page.

2) Terminal Emulation-based

By using the VT-100/ANSI compatible terminal emulation software, such as Microsoft HyperTerminal, user can configure VENUS 29xx SERIES via the Craft port with console cable. Table 3-1 lists some commonly used software, based on the type of PC platform you are using.

Table 3-1 Terminal Emulation Software

Operation System	Software
Windows 98/XP/NT/2000/2003/Vista	HyperTerminal (included with Windows software)
Linux	Minicom Or Kermit

Select the COM port used and setup the following settings:

- Speed: **115200 bps** (bit per second)

- Data Length: **8 bits**

- Parity Bit: **None**

- Stop Bit: **1 bit**

- Flow Control: **None**

3.1.1.2 Power ON VENUS 29xx Series

Power on your VENUS 29xx SERIES after the gateway is connected to your PC. It takes several seconds to perform the initialization and diagnosis. After the gateway has been finished initializing, the LED of service change to green. And now, you can connect to the gateway

3.1.1.3 Connect to VENUS 29xx Series

1)Web-based

The default parameters of VENUS 29xx Series Ethernet port are as follow:

Default IP address: 192.168.0.1

Default gateway: 192.168.0.254

Default net mask: 255.255.255.0

Before enter the web site of the VENUS 29xx Series you need to change the configuration of your PC network to the same network segment as the VENUS 29xx Series.

Open the web site by the web browser and you will see the login web page as shown in the following Figure.

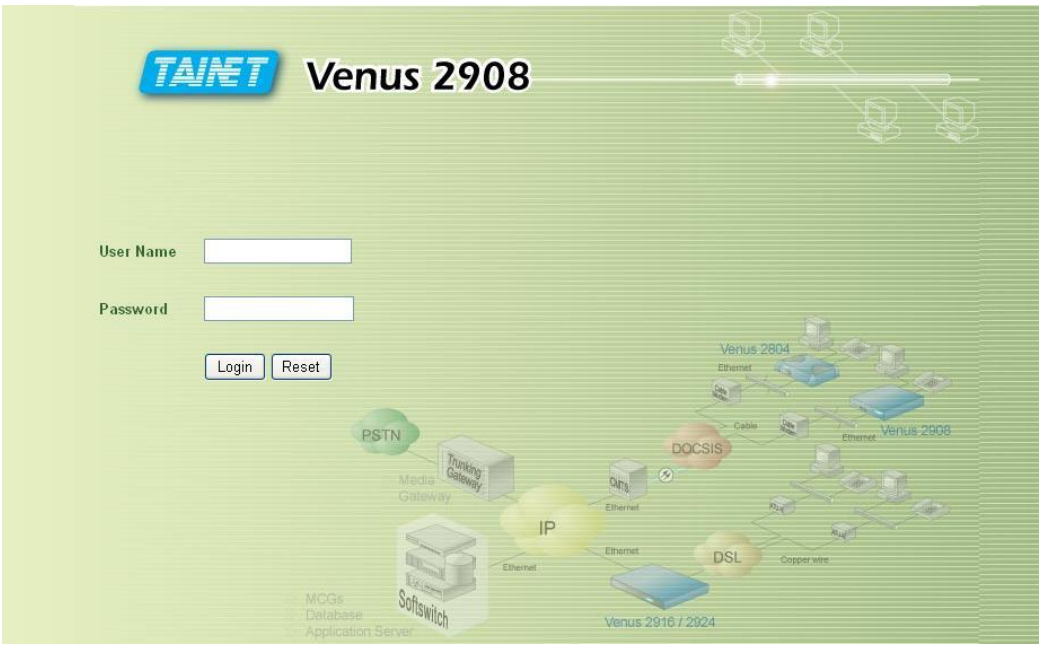


Figure 3-2 The Login Page

2) Terminal Emulation-based

You also can change the device network configuration by the Terminal Emulation instead of change your PC's.

After the Terminal Emulation startup press any key on the gateway. It will prompt the following messages:

```
**************************************************************
Sip version 1.00 compiled on Jun  3 2008 at 12:40:33
IAD 2908-8FXS  Hardware version 1.0
**************************************************************
Login:
```

Figure 3-3 The Login CLI Screen

3.1.1.4 Login

Username and Password are required to login the VENUS 29xx for protection from unauthorized access to the device.

Default User Name: admin

Default User Password: admin

1) Web-based

Type the User Name and the User Password, and then click the "Login" button go to the root menu as shown in the following Figure.

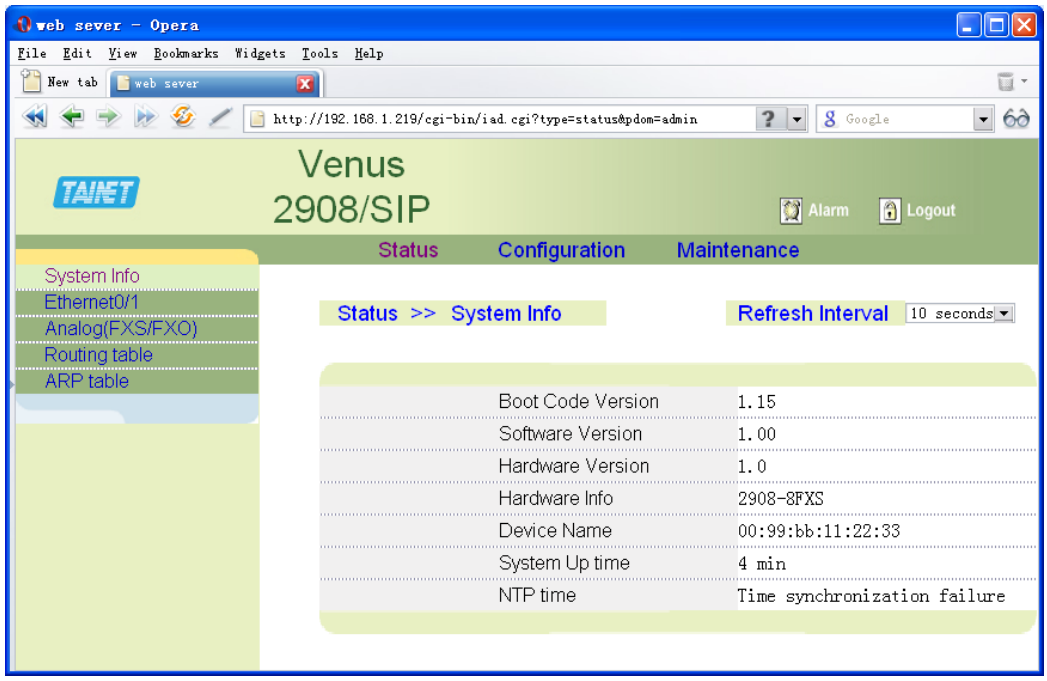


Figure 3-4 The root menu of the web site

There are three items, “Status”, “Configuration” and “Maintenance” in the top menu.

2) Terminal Emulation

When the security check is passed, the terminal UI displays the Root Menu as shown in the following Figure:

```
**********************************************************************
Sip version 1.00 compiled on Jun  3 2008 at 12:40:33
IAD 2908-8FXS  Hardware version 1.0
**********************************************************************
Login:admin
Password:
type help for more infomation.
admin>_
```

Figure 3-5 The Root Screen

3.1.1.5 Command List

Type the “help” in the Terminal Emulation when login successfully to show the Command List as shown in the following Figure:

```
admin>help
MXP Debug Application Command List
                Command Description
                      ? Displays Command Help Table
                   help Displays Command Help Table
---------------------------- User Commands
                    spy Set/Display Spy levels
                     st Status Commands
                   show Show System Configuration
                    set Set System Configuration
                     mt Maintenance Commands
                   save Save System Configuration
                 logout Logout System
```

Figure 3-6 The Command List Screen

3.1.1.6 Configure the Ethernet Interface

1) Set IP Address

Command format:

set interface ethernet <id(0..1)> ip_address <ip string for static>

Sample:

admin> set interface ethernet 0 ip_address 192.168.1.222

2) Set Default Gateway

Command format:

set interface ethernet <id(0..1)> default_gateway <ip string for static>

Sample:

admin> set interface ethernet 0 default_gateway 192.168.1.20

3) Save and run the setting

Type “save” to save and run the setting.

The two commands have been executed as shown in the following Figure.

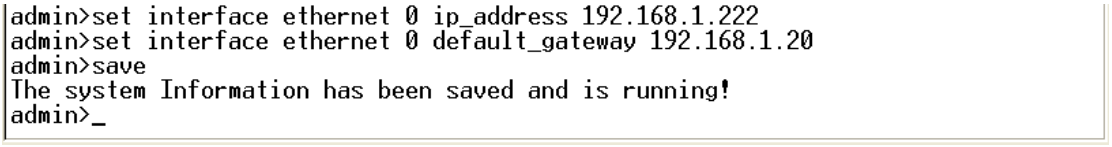

admin>set interface ethernet 0 ip_address 192.168.1.222
admin>set interface ethernet 0 default_gateway 192.168.1.20
admin>save
The system Information has been saved and is running!
admin>_

Figure 3-7 Set Ethernet Interface

3.1.1.7 Check the setting of Ethernet interface

Type “show interface Ethernet 0” in the Terminal Emulation to check the setting of Ethernet 0. The result of the status is shown as following.

admin>show interface ethernet 0

Ethernet0 config Parameters:

Interface mode:enable

IP mode:static

IP address:192.168.1.222

Net mask:255.255.255.0

Default gateway:192.168.1.20

DNS1:0.0.0.0

DNS2:0.0.0.0

DHCP use host name as Venus 2908 name:disable

DHCP use static dns:disable

DHCP trust server1:0.0.0.0

DHCP trust server1 net mask:0.0.0.0

DHCP trust server2:0.0.0.0

DHCP trust server2 net mask:0.0.0.0

PPPOE user name:

PPPOE password:

PPPOE use static dns:disable

PPPOE use remote gateway:enable

PPPOE reconnect when link down:enable

MTU size:1500

MAC address:

admin>

The Ethernet interface has modified and then you can complete the “Quick Start” configuration in the 6 steps, please refer to section 3.1.2.

3.1.2 Complete the “Quick Start” configuration

3.1.2.1 Start “Quick Start”

Click the “Configuration” of the top menu and come to “Quick Start” page. This page will guide you through completing the basic configuration of the VENUS 29xx Series as shown in the following Figure.

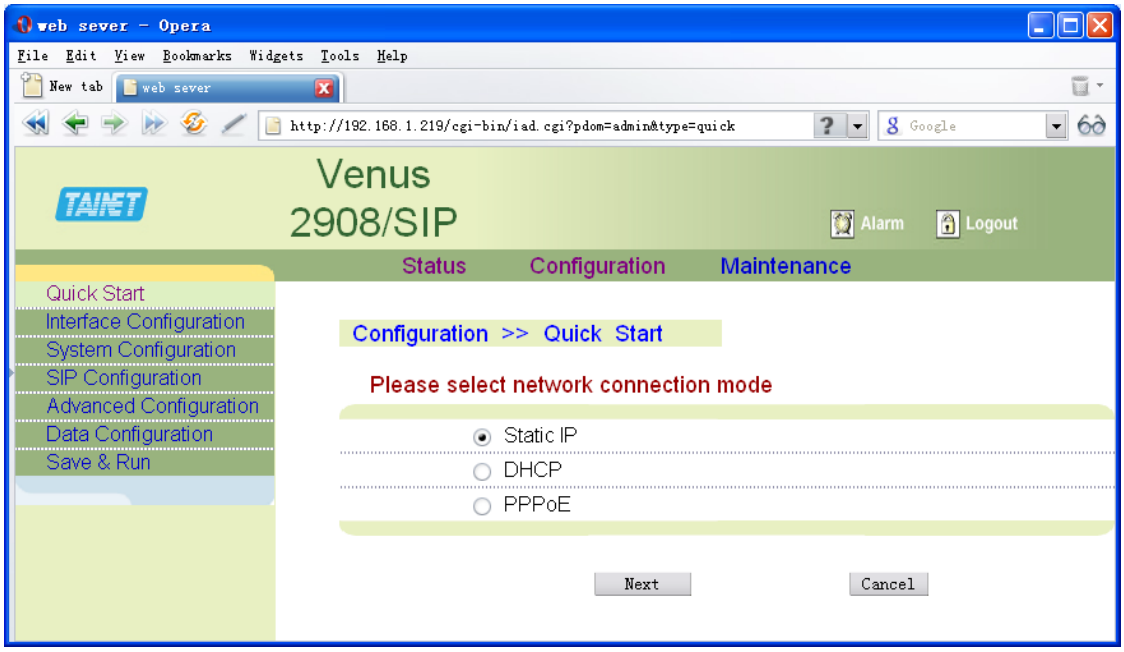


Figure 3-8 The Quick Start web page

3.1.2.2 Select Network Connection Mode

There are three modes for selecting the network, “Static IP”, “DHCP” and “PPPoE”.

Table 3-2 Network Connection Mode Configuration

Items	Description	Necessary
Network Connection Mode	Default:”Static IP”	Y

If you select “Static IP”, you need to set the VENUS 29xx Series network connection manually. This is the recommended way for the configuration. The “DHCP” mode and “PPPoE” mode are provided to advanced user. The web page of Network Connection Mode configuration is shown in the following Figure.

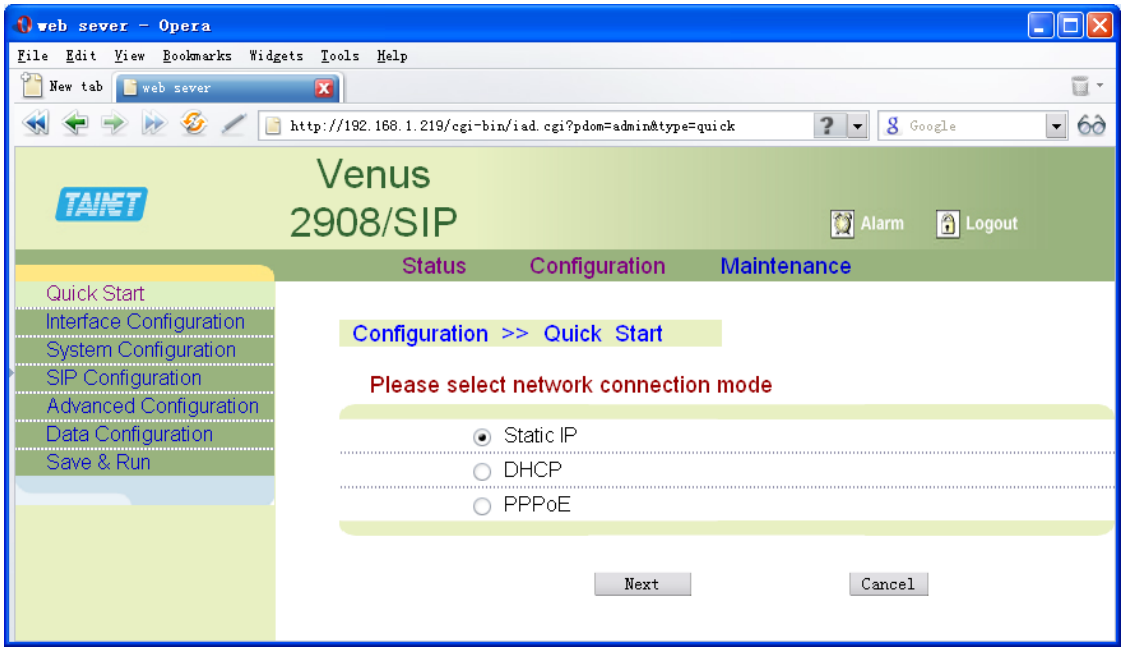


Figure 3-9 Network Connection Mode configuration

Caution:

If “DHCP” is selected, the detailed “DHCP Configuration” can only be done after this whole “Quick Start” setting is completed.

Click the “Next” button to continue.

3.1.2.3 Static IP setting

If you set the SIP proxy and the registrar server with its domain name but not its IP address, you need to set the DNS server IP address. You may set the second DNS server if you have two. The second DNS server is the backup of the first one.

Table 3-3 Static IP Configuration

Items	Description	Necessary
IP address	The local IP address of the VENUS 29xx.	Y
Net mask	The net mask of the local IP	Y
Default gateway	The default gateway to route packages for other network.	Y
DNS1 address	The DNS server IP address from which VENUS 29xx can get ip information when domain name is used.	N
DNS2 address	The backup DNS server IP address.	N

The Static IP configuration web page is shown in the following Figure.

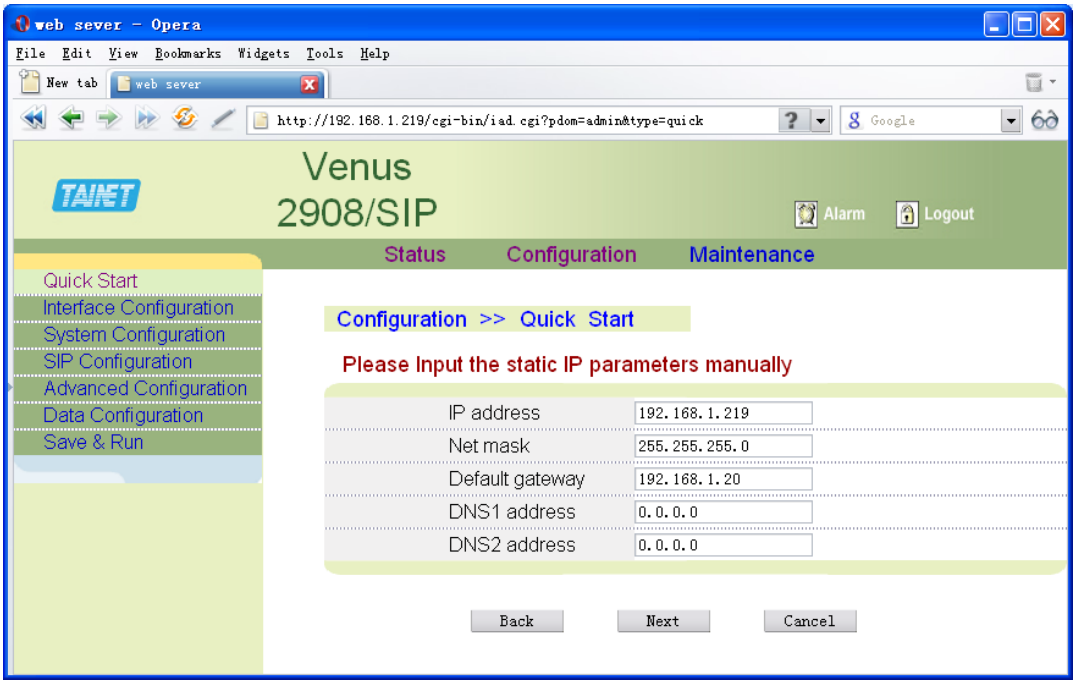


Figure 3-10 Static IP Configuration

Click the “Next” button to continue.

3.1.2.4 Sip Server setting

You need to set the SIP proxy and the register server manually. Sometime, they will share the same configuration if they are combined in a same system.

Table 3-4 SIP Server Configuration

Items	Description	Necessary
Domain Name	The domain name of the Proxy Server.	N
Proxy address	IP address or domain name of the softswitch. No default.	Y
Proxy Port	Port of the soft switch, the default value is 5060 .	N
Registrar address	IP address or domain name of the registrar server. No default.	Y
Registrar Port	Port of the registrar server, the default value is 5060.	N
Outbound Proxy	The IP address or domain name of the	N

Items	Description	Necessary
address	outbound proxy	
Transport way	The transport of SIP signaling, the default value is UDP.	Y

The SIP Server configuration web page is shown in the following Figure.

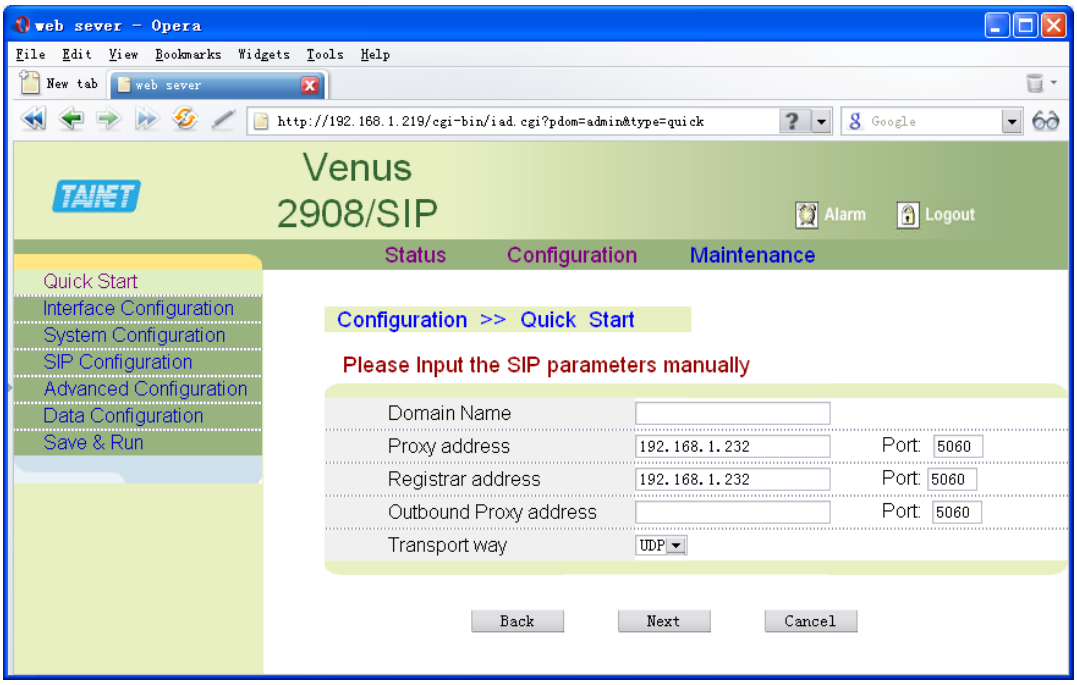


Figure 3-11 SIP Server Configuration

Caution:

You may set the “Proxy address”, the “Registrar address” and the “Outbound Proxy address” with its domain name if the DNS server is configured.

Click the “Next” button to continue.

3.1.2.5 FXS/FXO setting

We supply you default configuration of the FXS/FXO. You can change the User Number and User authentication ID parameters of Ports as needed.

Table 3-5 FXS/FXO Configuration

Items	Description	Necessary
User Number(Port1)	Default: 1001.	Y
User Number(Port2~8) or (port2~32)	Default: increment;	Y
User authentication ID(Port1)	No default	N
User authentication ID(Port2~8) or (port2~32)	Default: Same;	N
User authentication password(Port1)	No default	N
User authentication password(Port2~8) or (port2~32)	Default: Same;	N
Local server port number	Default: 5060;	N
Register Mode	Default: Enable; “Shut Down”- The port can do nothing. “Disable”-The port only allowed to receive call. “Enable”-The port work normal.	Y

The FXS/FXO configuration web page is shown in the following Figure.

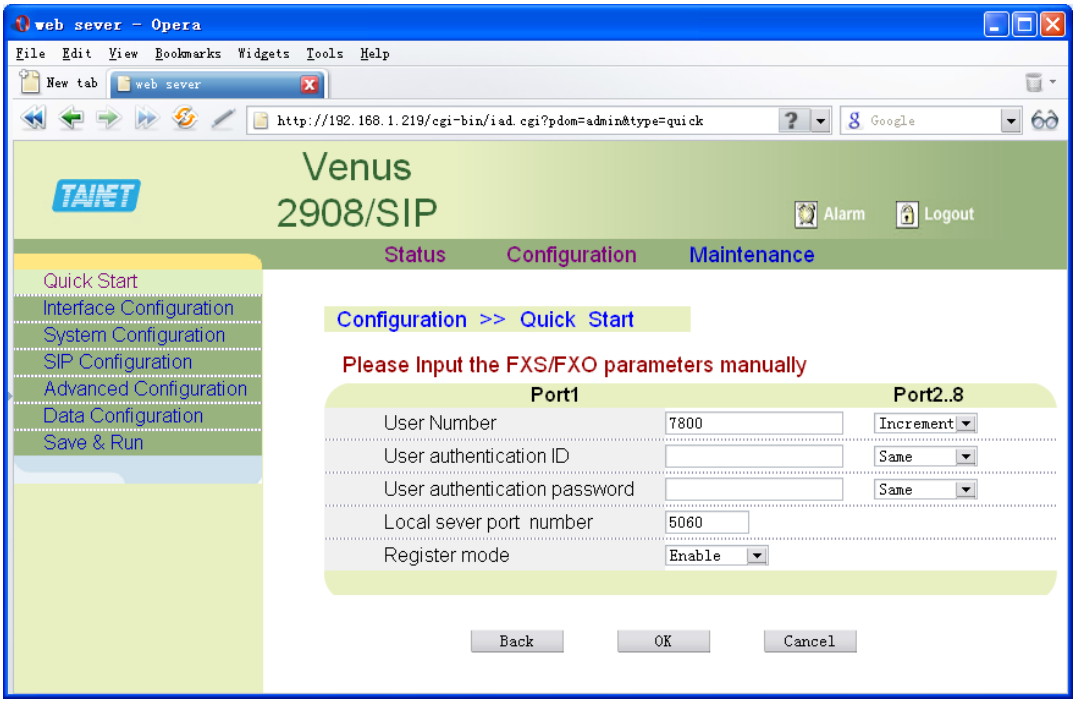


Figure 3-12 FXS/FXO Configuration

You need to set the “User authentication ID” and the “User authentication password” when the Sip proxy and the register require authentication.

Click the “OK” button and you come to the last page of the “Quick start” which shows you “Save and Run the web server configuration Succeed!”. All the ports of the Venus 29xx Series will register to the server automatically after this operation.

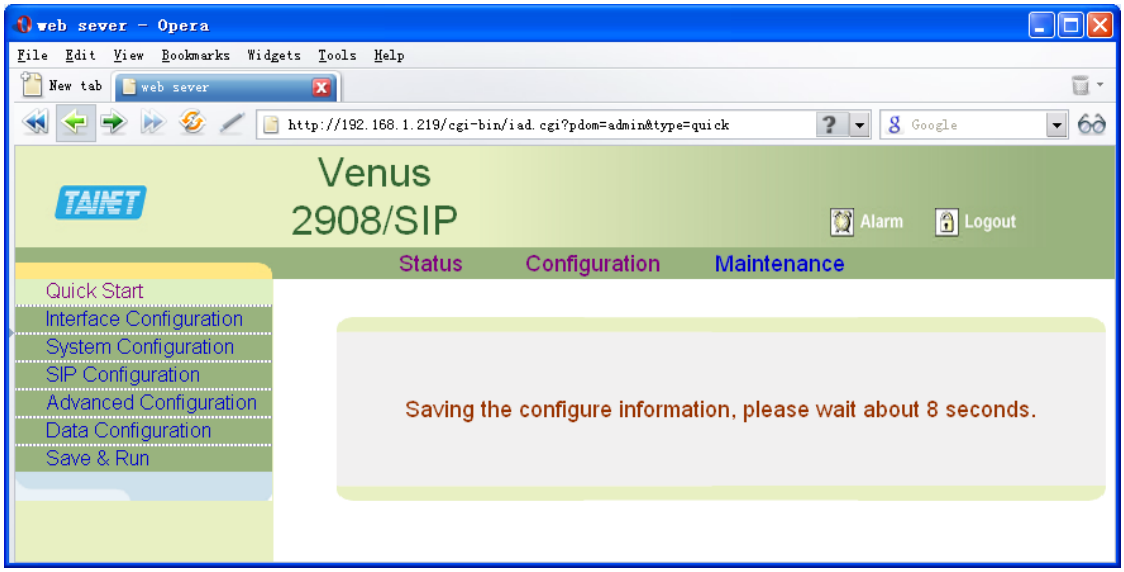


Figure 3-13 Saving the configure information

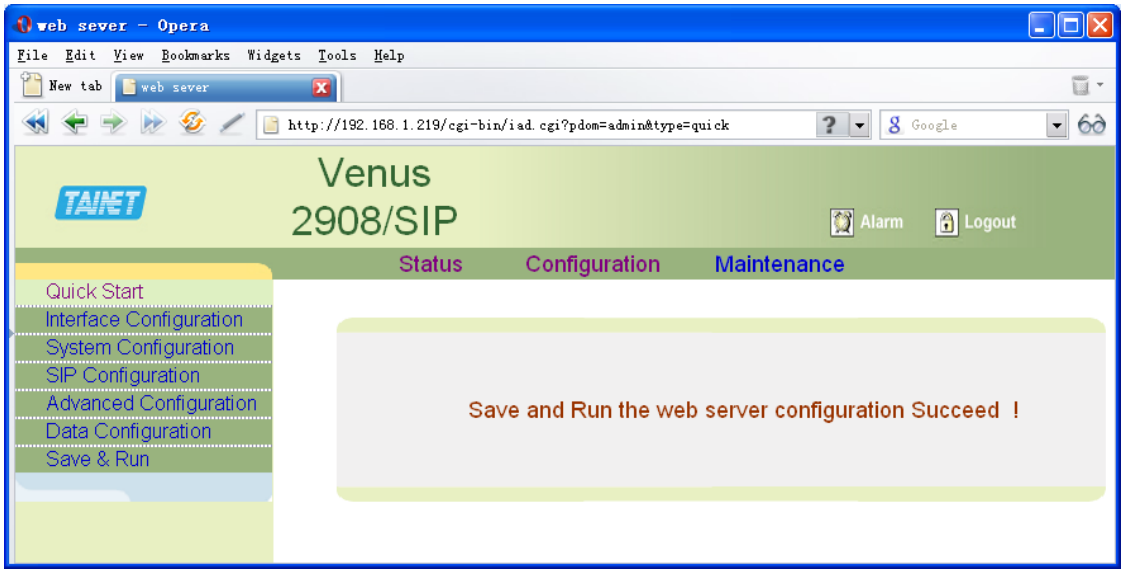


Figure 3-14 Quick Start completed

3.1.2.6 Check the FXS/FXO state

Now, you have completed the Quick Start configuration of the VENUS 29xx Series.

You can click the “status” of the top menu and select the “Analog (FXS/FXO)” of the left panel to check the result of this process. If you see “Register” state of the ports are all “OK”, the configuration is successful.

The FXS/FXO state is shown in the following Figure.

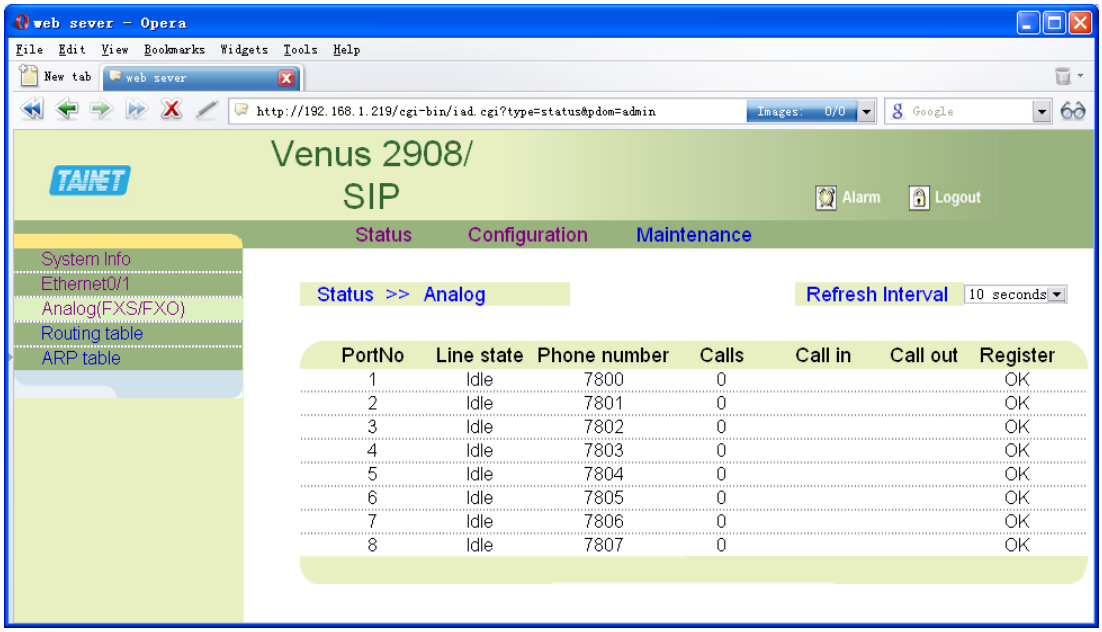


Figure 3-15 Check FXS/FXO state

Now, the Venus 29xx Series is ready for VoIP service.

3.2 Quick Setup

3.2.1 Login

Setup PC's IP address to be using the same network segment as Venus 29xx Series and directly connect PC to the WAN port of Venus 29xx Series. Enter the default IP address http://192.168.0.1 to access the Web user interface. Default user name is "admin" and password is "admin".

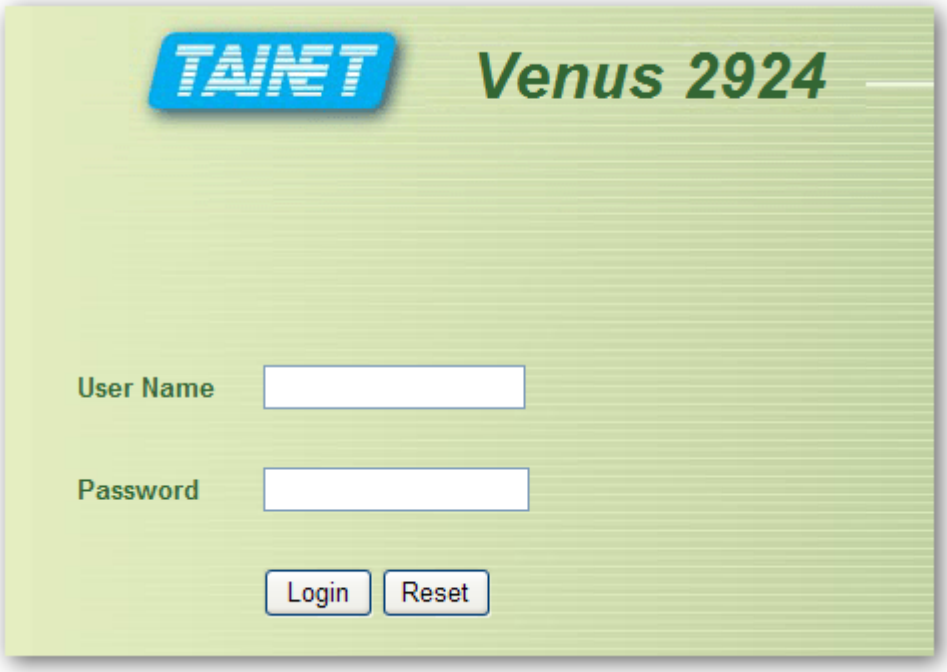


Figure 3-16 Login Page

3.2.2 Main Page

After Login, user will see the main page of Venus 29xx Series.

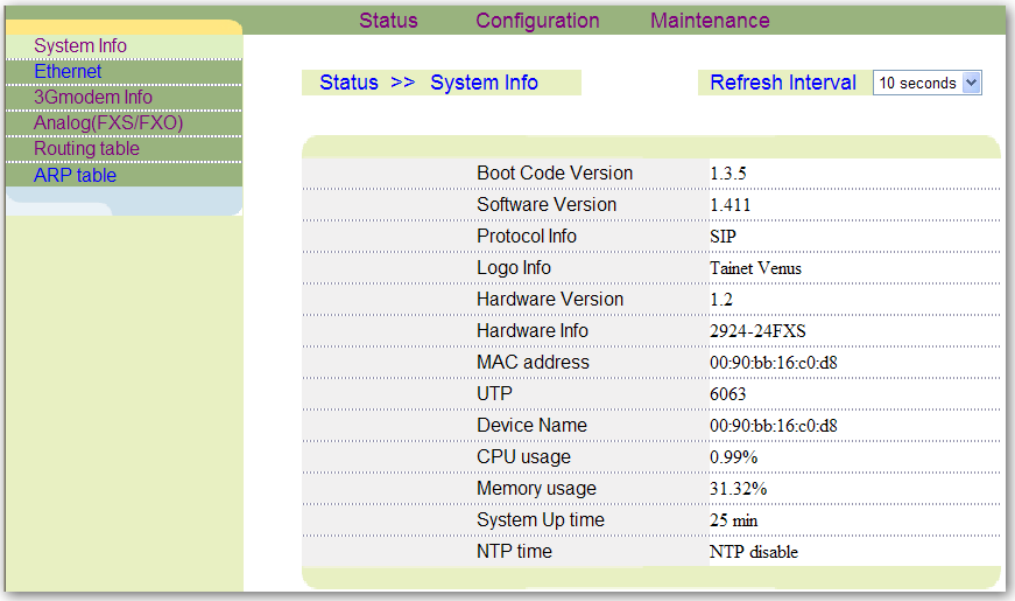


Figure 3-17 Main Page

3.2.3 Network Setting

Users now will need to change the network configurations to comply with field application. Select **Configuration >> Interface Configuration >> Ethernet** to setup desired IP address, Net mask and Default Gateway or select “DHCP” or “PPPOE” for Venus to automatically derive IP setting from the server. Select “OK” to confirm settings and select “Save & Run” to apply settings.

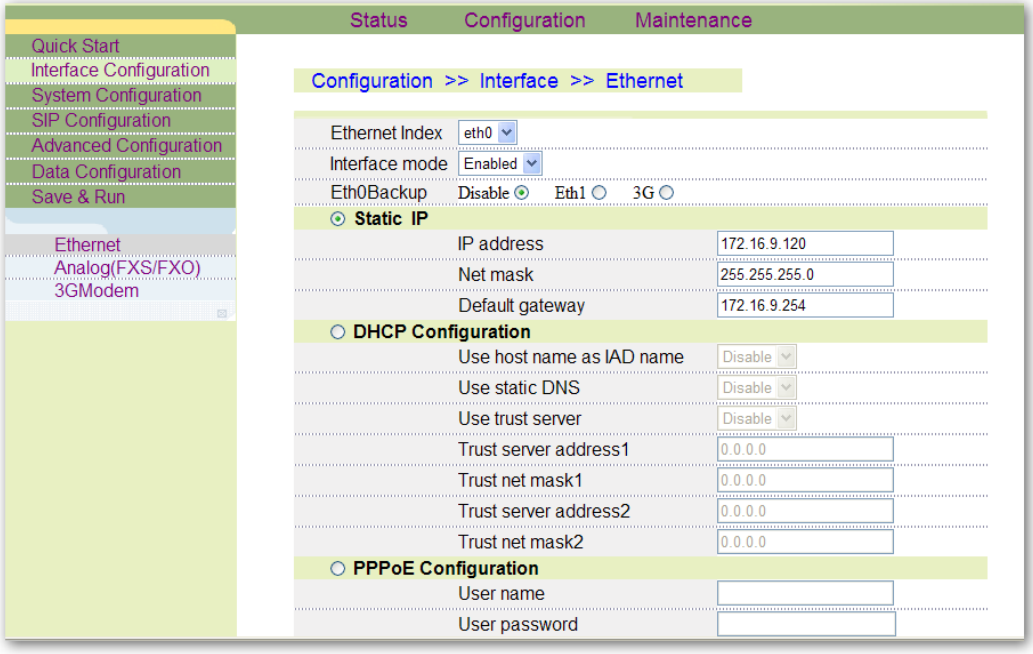


Figure 3-18 Network Configuration

3.2.4 SIP Setting

In order for Venus to be successfully registered to the SIP server, users need to enter the SIP proxy and registration addresses. Click **Configuration >> SIP Configuration >> Realm** to configure Venus SIP realm. Select realm index to 0 and change realm valid to “Enable”. Click “OK” to confirm and select “Save & Run” to apply.

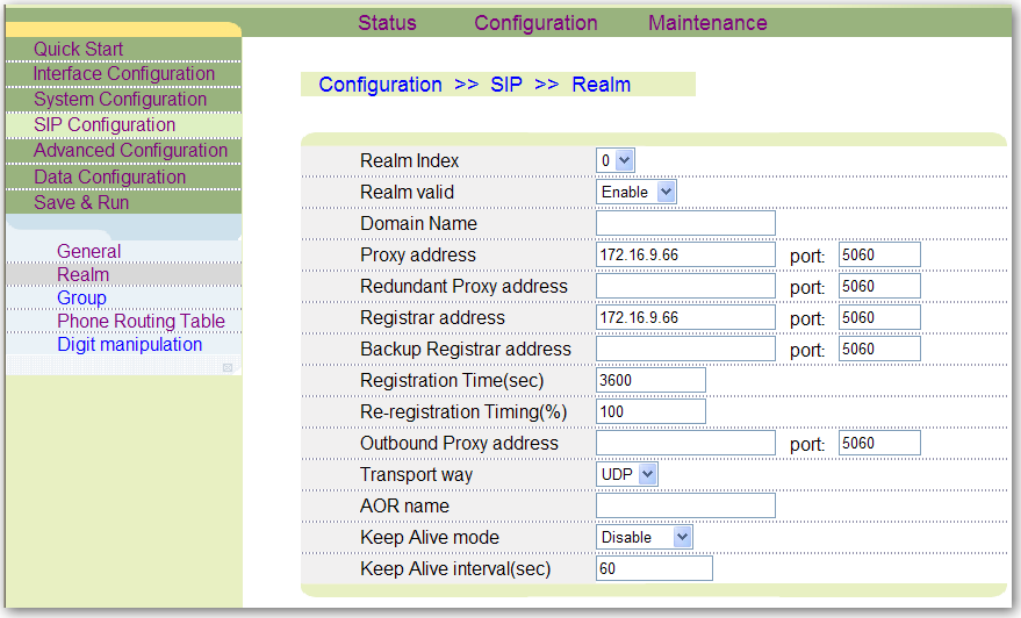


Figure 3-19 SIP Realm

3.2.5 Analog (FXS/FXO) Setting

Setup user number and authentication ID and password for each analog FXS or FXO ports to have them successfully registered to SIP server. Select register mode to “Enable” and enter correct user number and authentication ID and password. Select “OK” to confirm and select “Save & Run” to apply.

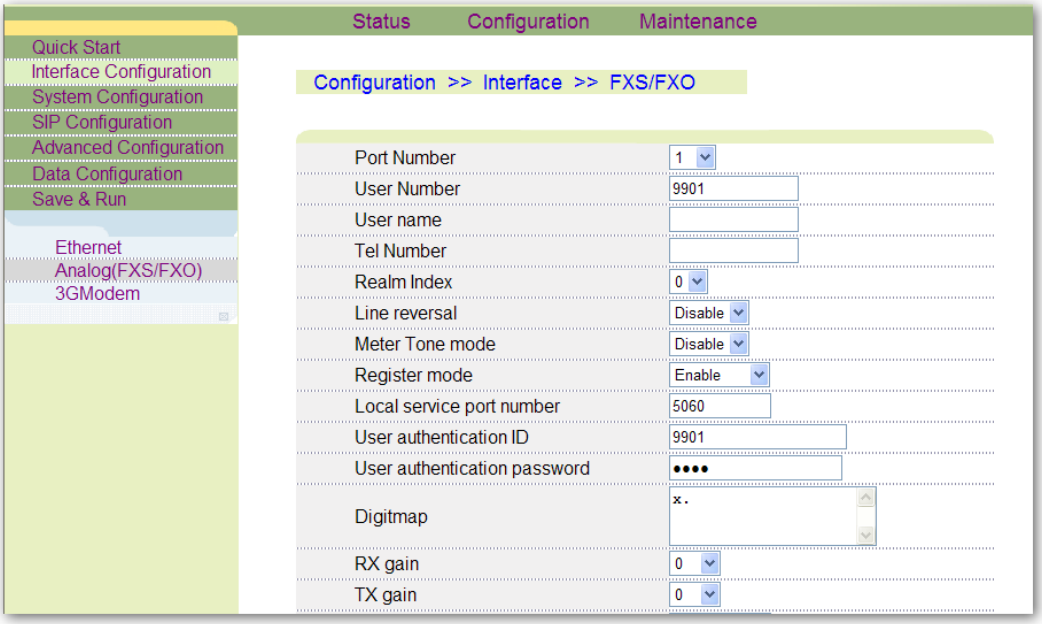


Figure 3-20 Analog (FXS/FXO) settings

3.2.6 Status Check

After all configurations are done, user can check the current registration status at **Status >> Analog (FXS/FXO)**.

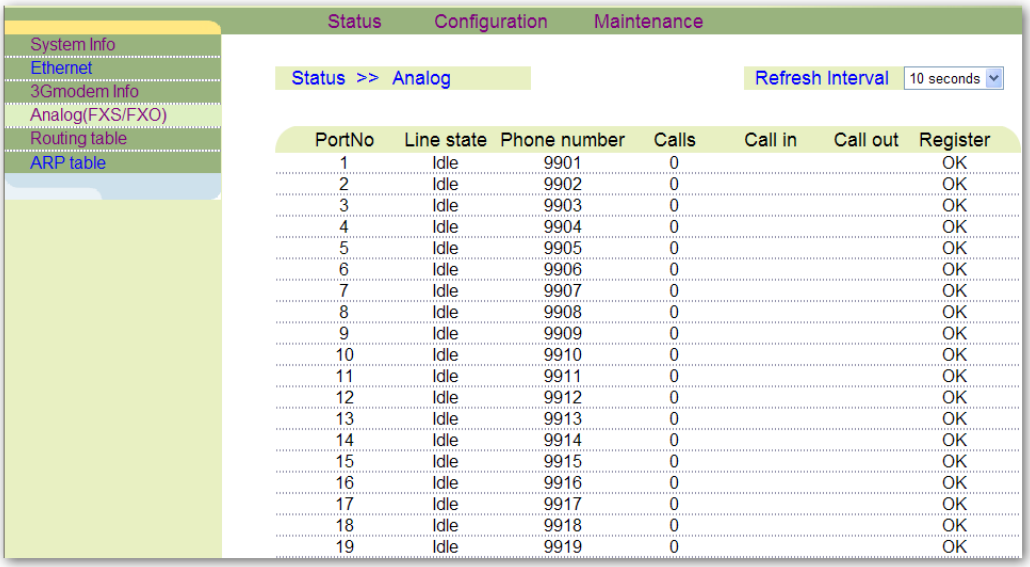


Figure 3-21 Status Check

3.3 Login and Logout

There are three types of user accounts built in the Venus 29xx Series: Administrator, Operator and Monitor. Each type of user can change its own name and password.

Table 3-6 VENUS 2908 User

User Class	User Name	Default Password	Operating
Administrator	admin	admin	System manager. Full control.
Operator	user		Only allow Ethernet configuration and monitor.
Monitor	guest		Only allow monitor.

1) Web-based

After input the right user name and password, click “Login” button to log in the web. Click “Logout” icon of the main page to log out the web.

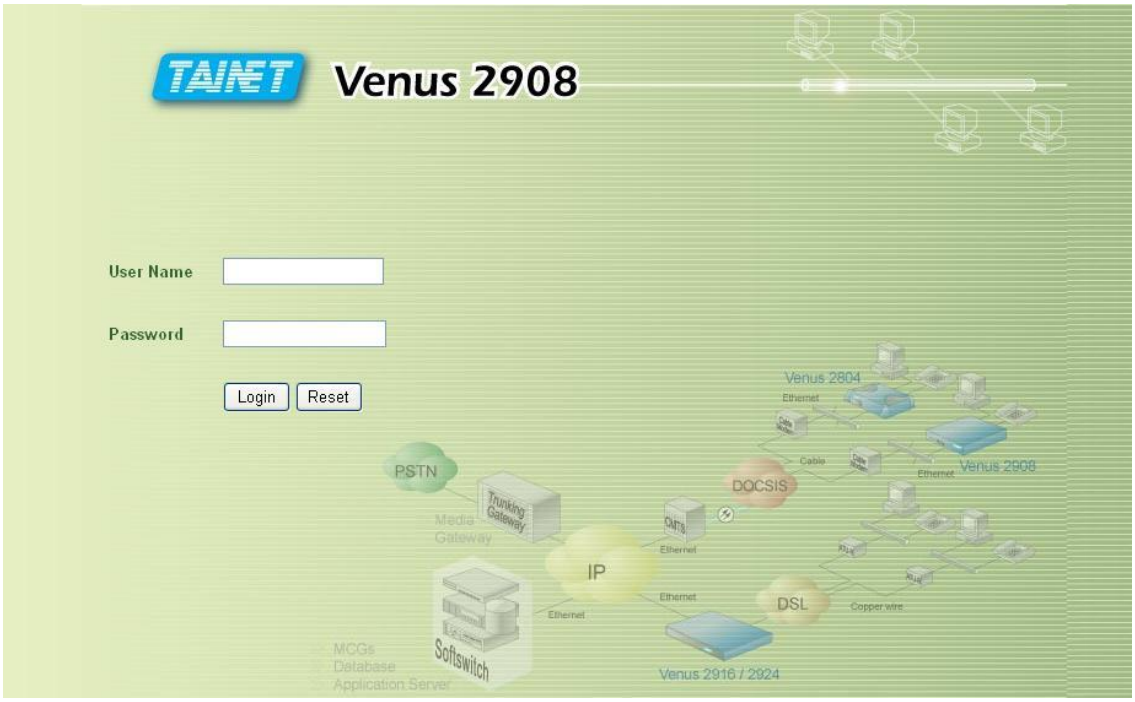


Figure 3-22 Web-based Login interface

2) Terminal Emulation-based

Input the right user name and password and then press “enter” key to login the CLI

interface. Input “logout” and then press “enter” key to logout the CLI interface.

```
**************************************************************
Sip version 1.00 compiled on Jun  3 2008 at 12:40:33
IAD 2908-8FXS  Hardware version 1.0
**************************************************************
Login:
```

Figure 3-23 Terminal Emulation-based Login interface

3.4 Navigating the Web Interface

The following Figure shows the style of the Web UI. Indications point out the layout of the UI and help user to navigating through the User Interface.

There are three categories in the main menu, “Status”, “Configuration” and “Maintenance”. Each category has several sub-menu also for special functions.

Status

To get system info and status working state;

Configuration

To set and update system configuration;

Maintenance

To manage the system fault, control the device and diagnostic.

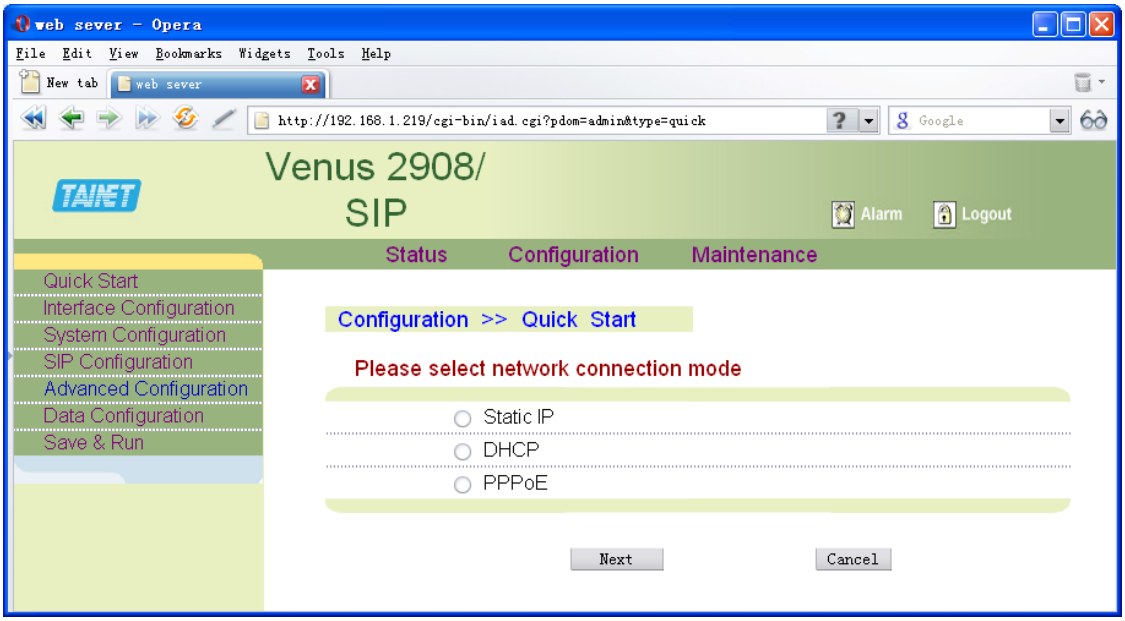


Figure 3-24 Web UI Main Page

Table 3-7 Main Menu Tree

Menu	Sub-Menu	Description
Status	System Info	Show the version of the Software, Hardware, Boot Code and etc.
	Ethernet	Show the configuration of the eth0 and eth1. (eth2~4 for 2916/2924/2932).
	3Gmodem Info	Show current status of 3G modem connection.
	Analog (FXS/FXO)	Show the state of each FXS/FXO.
	Routing table	Show the routing table.
	ARP table	Show the ARP table.
Configuration	Quick Start	A guide for user to start the system quickly.
	Interface Configuration	Configure Ethernet, Analog, 3GModem and PSTN.
	System Configuration	Configure (System) General, System Time, User Management, UI Management, Provision ,SNMP, TR069, FM Threshold and Syslog
	SIP Configuration	Configure (SIP) General, Realm, Group, Phone Routing Table and Digit manipulation.

Menu	Sub-Menu	Description
	Advanced Configuration	Configure RTP Management, Calling Feature, Region Profile, DDNS, NAT Traversal, SLIC Configuration and Meter Configuration.
	Data Configuration	Configure Static Routing, NAT Mapping, DHCP Sever, QoS and VLAN Management. Note: Static Routing and NAT Mapping are displayed if interface eth1 is in router mode.
	Save & Run	Save all the changes and run them.
Maintenance	Fault Management	FM Monitoring and Clear FM.
	Performance Management	Ethernet, Call Control, RTP, NAT, DHCP Server, SIP Agent, SIP Call Initiation and Clear. Notes: NAT && DHCP server are displayed if interface eth1 is in router mode.
	Diagnostics	Tools including Ping, STUN, Traceroute.
	Device Control	F/M Upgrade, Config Up/Down, Syslog Upload, Reset to Default and Reboot.

3.5 Configuration

Beside “Quick Start” and “Save & Run”, the Configuration menu is divided into seven groups, “Quick Start”, “Interface Configuration”, “System Configuration”, “SIP Configuration”, “Advanced Configuration”, “Data Configuration” and “Save & Run”.

“Quick Start” is used to help user starting VENUS 29xx quickly, please refer to Section 3.1 for more information.

Table 3-8 The Configuration Menu Items

Menu	Description
Quick Start	A guide for user to start the system quickly.
Interface Configuration	Configure Ethernet, Analog, 3GModem and PSTN.

Menu	Description
System Configuration	Configure (System) General, NTP, User Management, UI Management, Provision ,SNMP, TR069, FM Threshold and Syslog
SIP Configuration	Configure (SIP) General, Realm, Group, Phone Routing Table and Digit Manipulation.
Advanced Configuration	Configure RTP Management, Calling Feature, Region Profile, DSP Allocation, DDNS, NAT Traversal, SLIC Configuration and Meter Configuration.
Data Configuration	Configure Static Routing, NAT Mapping, DHCP Sever, QoS and VLAN Management.
Save & Run	Save all the changes and run them.

3.5.1 Save & Run

"Save & Run" is not a single group of Configuration. Save the confirmed changes of configuration and run them. If no change has been confirmed, do nothing.

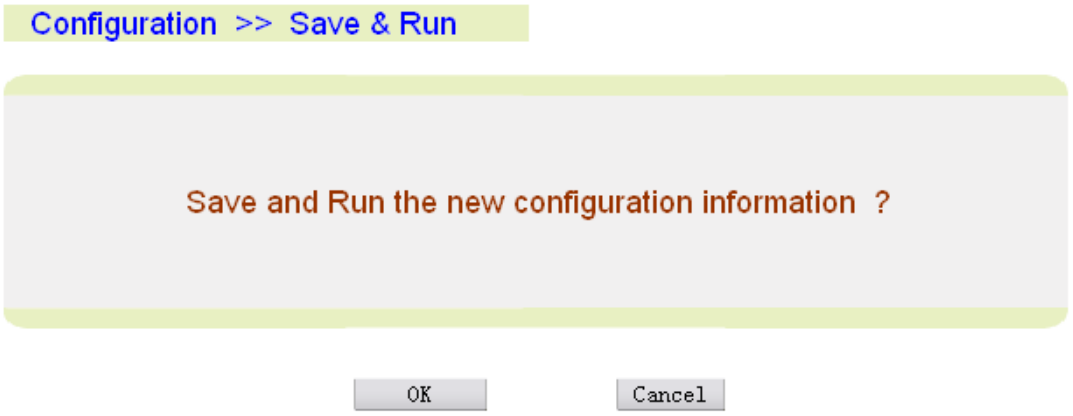


Figure 3-25 Prompt of "Save & Run"

Click "OK" button to save and run or click "Cancel" button to give up.

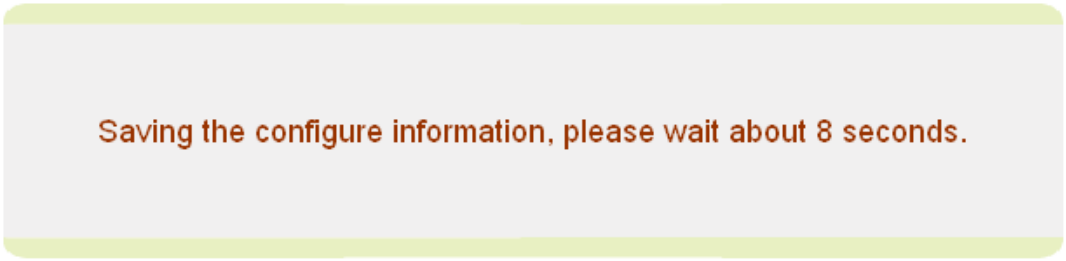


Figure 3-26 Prompt after click "OK" button on "Save & Run" page.

3.5.2 Interface Configuration

Table 3-9 The Interface Configuration Menu Items

Menu	Description
Ethernet	Configuration of the eth0, eth1 or eth2~4(2916/2924/2932) such as “Static IP”, “DHCP”, “PPPoE”, “DNS” and so on.
Analog(FXS/FXO)	Configuration of the port1~8(2908), port1~16(2916), port1~24(2924) or port1~32(2932).
3GModem	3G Modem configuration.
PSTN Lifeline (2908 8S Model only)	PSTN configuration.

3.5.2.1 Ethernet Configuration

There are two Ethernet interfaces in VENUS 2908 and five Ethernet interfaces in VENUS 2916/2924/2932. Ethernet interface 0 is usually called WAN. Ethernet interface 1~4 can be considered as LAN port. Ethernet interface 0 supports three methods of obtaining the IP address: static, DHCP or PPPoE. Ethernet interface 1 operates on four modes: disable, router, switch and Eth0 backup. Ethernet 2 to 4 operates on two modes: switch and disable.

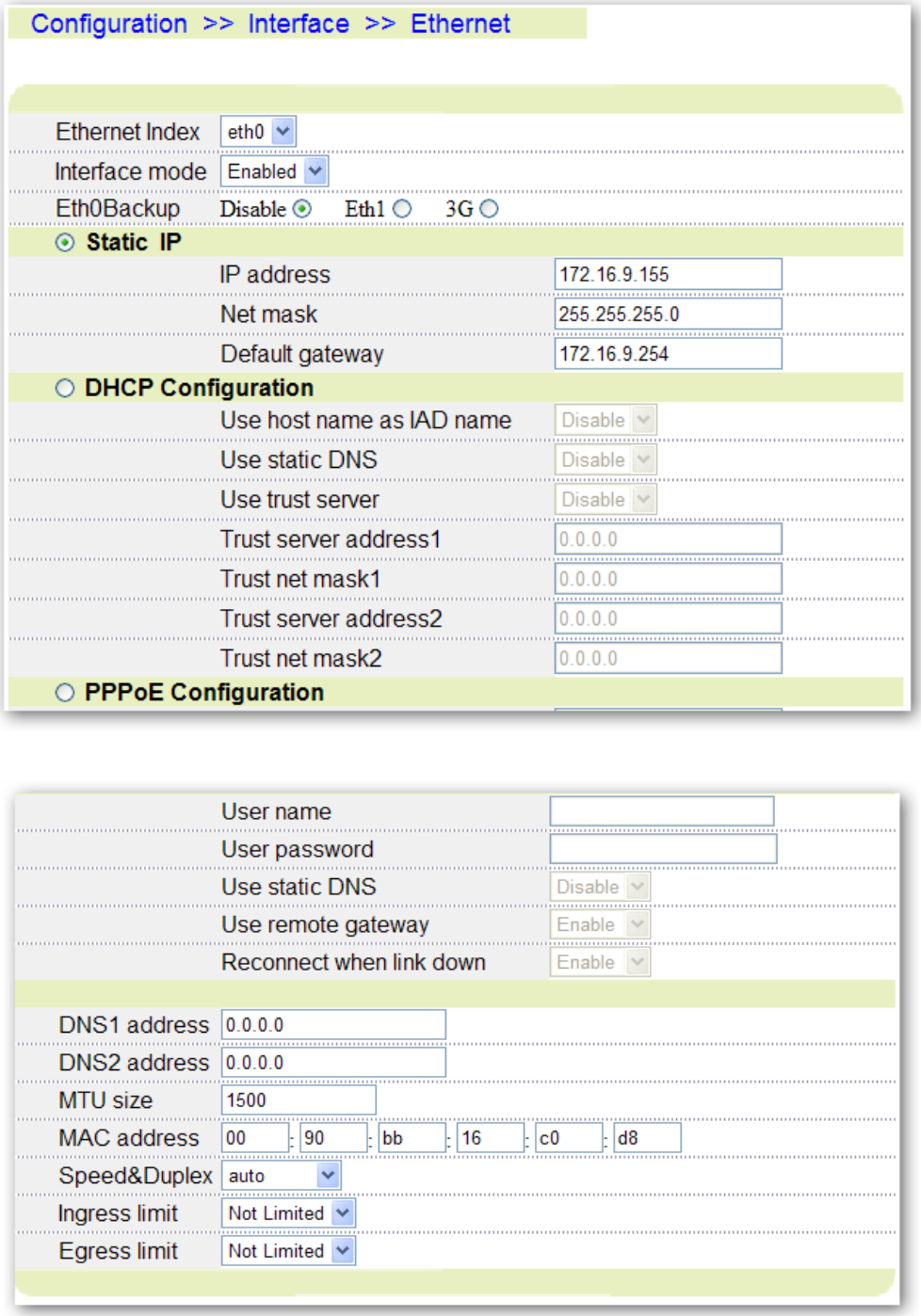


Figure 3-27 Ethernet Configuration

Table 3-10 Description of Ethernet Configuration Items

Field	Description
Interface mode	Interface working mode, for interface 0, it should be always Enabled. For interface 1, it can be configured to “Disable”, “router” or “switch” or “Eth0 backup”. For interface 2~4, it can be configured to “Disable” or “switch”.
Eth0Backup	Backup interface for Eth0: 1.Disable 2.Eth1 3.3G
IP mode	Quick Start parameter. There are three IP mode: 1.Static IP address (default) - IP is assigned to a specified one; 2.DHCP client mode - Auto assigned by DHCP server; 3.PPPoE mode - Auto assigned by PPPoE server;
IP address	Quick Start parameter The static IP address of specified interface.
Net mask	Quick Start parameter The net mask of specified interface.
Default gateway	Quick Start parameter The default gateway address of specified interface.
DNS1 address	Quick Start parameter The DNS server 1 IP address
DNS2 address	Quick Start parameter The DNS server 2 IP address
DHCP use host name as IAD name	1.Disable(default); 2.Enable;use host name in DHCP protocol
DHCP use static DNS	1.disable(default); 2.enable; ignore the DNS information in DHCP offer.
DHCP use trust server	1.disable(default); 2.enable
DHCP trust server address 1	The first trust DHCP server network address.
DHCP trust net mask 1	The first trust DHCP server net mask.
DHCP trust server	The second trust DHCP server network address.

address 2	
DHCP trust net mask 2	The second trust DHCP server net mask.
PPPoE user name	The user name of PPPoE authentication.
PPPoE password	The password of PPPoE authentication.
PPPoE use static DNS	1.disable(default); 2.enable; ignore the DNS information in PPPoE offer.
PPPoE use remote gateway	1.disable; 2.enable(default);use this gateway as VENUS 2908 default gateway
PPPoE reconnect when link down	1.disable; 2.enable(default);
MTU size	The MTU size of specified interface
MAC address	The MAC address of specified interface, configurable by admin user only.
Speed&Duplex	0 = auto (default) 1 = 10M full 2 = 10M half 3 = 100M full 4 = 100M half
Ingress limit	Set up ingress limitation for incoming packets
Egress limit	Set up egress limitation for outgoing packets

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

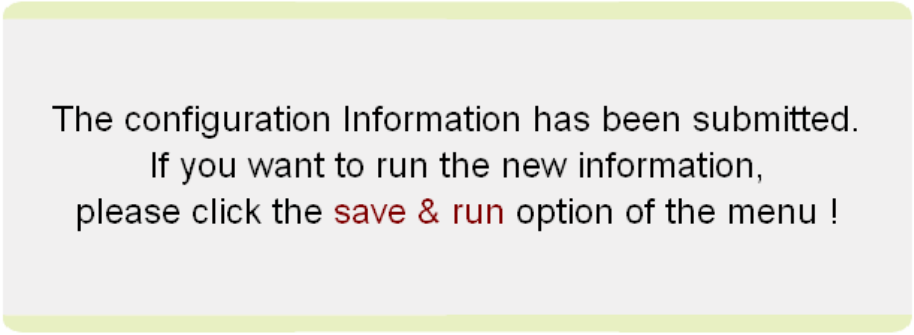

The configuration Information has been submitted.
If you want to run the new information,
please click the save & run option of the menu !

Figure 3-28 Prompt after click “OK” button

3.5.2.2 Analog (FXS/FXO) Configuration

The analog line means either FXS interface or FXO interface. Generally the total number is different for each model. The configuration parameter is specified by each interface (Port Numbers 1 to 8 or 1 to 32).

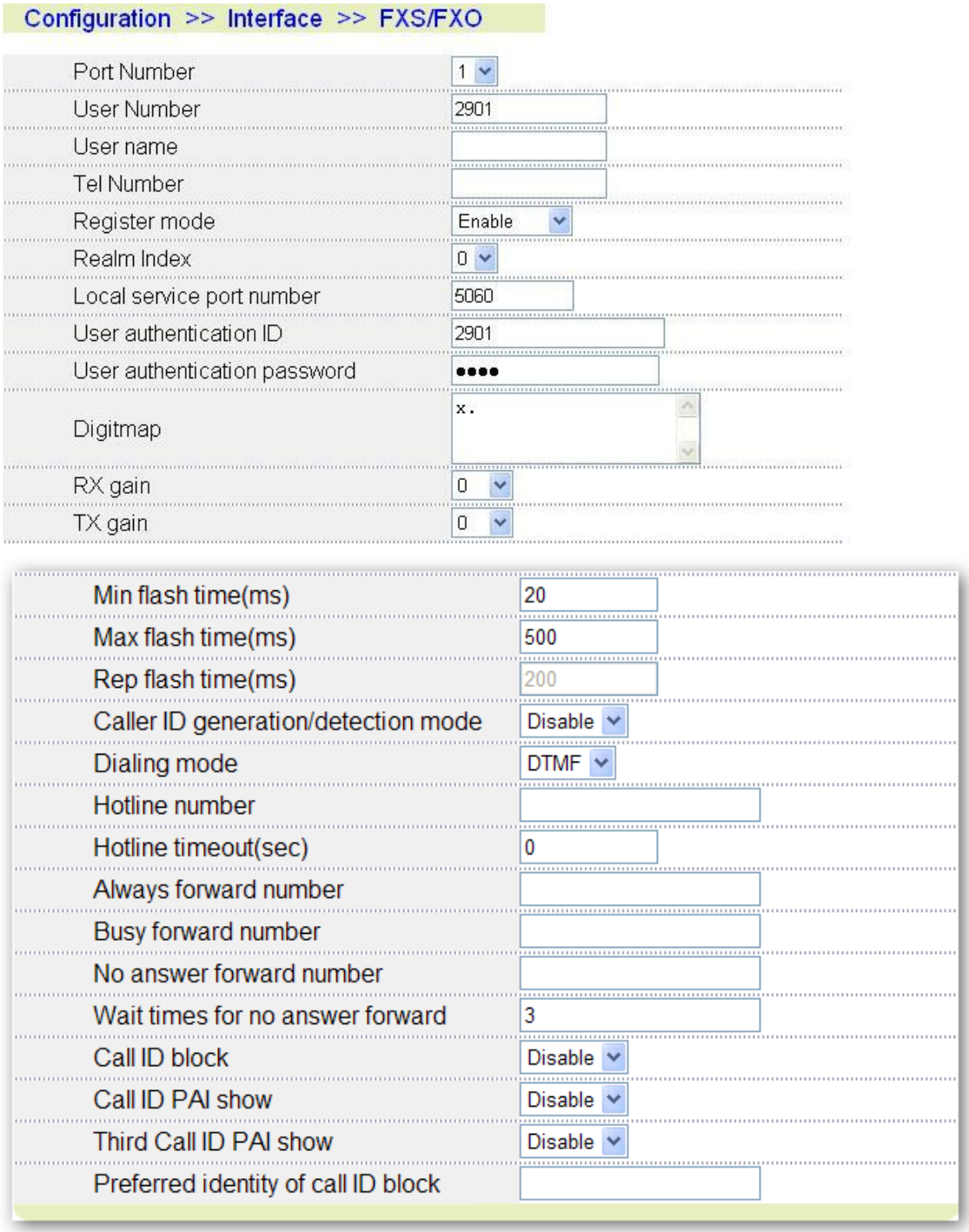


Figure 3-29 Analog (FXS/FXO) Configuration

Table 3-11 Description of Analog (FXS/FXO) Configuration Items

Field	Description
Port Number	Port number from 1 to 8(2908) or 1 to 16(2916) or 1 to 24(2924) or 1 to 32(2932).
User Number	Quick Start parameter (Same or increment) The register numbers which allow using the same number for different port. The ports used the same number is taken as group binding number.
User name	In SIP, it presents the display name for specified user.
Tel Number	User number for Tel format
Register mode	Quick Start parameter. Enable/Disable register to server. 1.Shut down 2.Disable(default) 3.Enable
Realm Index	Quick Start parameter. The assigned realm index (0..7). Assume the realm 0 always exists.
Local service port number	Quick Start parameter. Local service port number. Default is 5060 for SIP
User authentication ID	Quick Start parameter (Same or increment) The authentication user ID.
User authentication password	Quick Start parameter (Same or increment) The authentication user password.
Digitmap	The default digitmap for collect digits
RX gain	The RX gain. (-20..20)
TX gain	The TX gain. (-20..20)
Min flash time	The minimum flash time
Max flash time	The maximum flash time
Rep flash time (2908 4S4O and 2932 FXO)	The report flash time
Line reversal	1.Disable 2.Enable(Default)(Send line reversal signal for FXS port and

	detect line reversal signal for FXO)
Caller ID generation/detection mode	1.Disable(Default); 2.Enable;
Dialing mode	1.Both 2.DTMF 3.Pulse
Hotline number	The hotline number or URL
Hotline timeout	The time to wait for digits input. If time out do hotline.
Always forward number	The forward number or URL for always forward.
Busy forward number	The forward number or URL for busy forward.
No answer forward number	The forward number or URL for no answer forward.
Wait times for no answer forward	The ring times to wait before No answer forward to configure number.
Call ID block	1.Disable(Default) 2.Enable The Asserted ID mode defines the header that is used in the generated INVITE request. The header also depends on the calling Privacy: allowed or restricted. The P-asserted (or P-preferred) headers are used to present the originating party's Caller ID. The Caller ID is composed of a Caller Number and (optionally) a Caller Name. P-asserted (or P-preferred) headers are used together with the Privacy header. If Caller ID is restricted, the 'Privacy: id' is included. Otherwise for allowed Caller ID the 'Privacy: none' is used. If Caller ID is restricted (received from Tel or configured in the gateway), the From header is set to <anonymous@anonymous.invalid>.
Call ID PAI show	1.Disable(Default); 2.Enable;
Third Call ID PAI show	1.Disable(Default); 2.Enable;
Preferred identity of call ID block	Preferred identity in call ID block mode

If the configuration is finished, click "OK" button bellow the web page to confirm changes or click "Cancel" button to deny it.

3.5.2.3 3GModem

This page is used to set up configurations for 3G modem connection

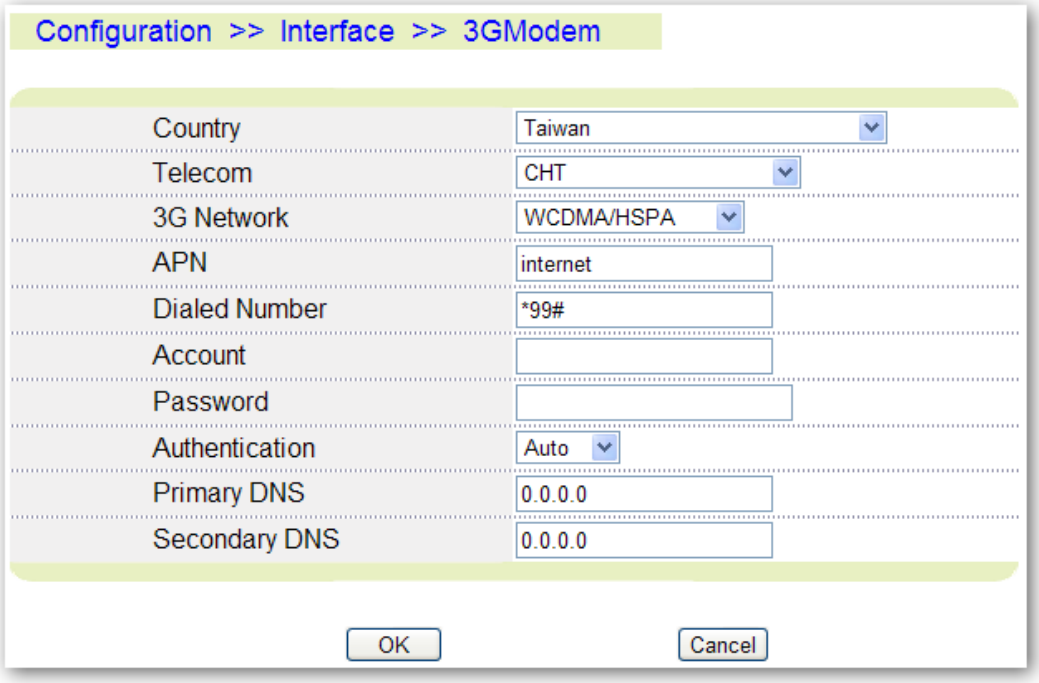


Figure 3-30 3G Modem

Table 3-12 Description of 3G Modem Configuration Items

Field	Description
Country	Country of 3G modem network
Telecom	Telecom of 3G modem network
3G Network	Network type to be used in 3G modem connection
APN	Access Point Name of 3G modem network
Dialed Number	Number dialed to establish 3G modem connection
Account	Account of 3G modem
Password	Password of 3G modem

Field	Description
Authentication	Authentication type : 1.Auto (Default) 2.PAP 3.CHAP
Primary DNS	Primary DNS used in 3G modem network
Second DNS	Secondary DNS used in 3G modem network

3.5.2.4 PSTN Lifeline Configuration

This configuration is available in Venus 2908-8S model only.

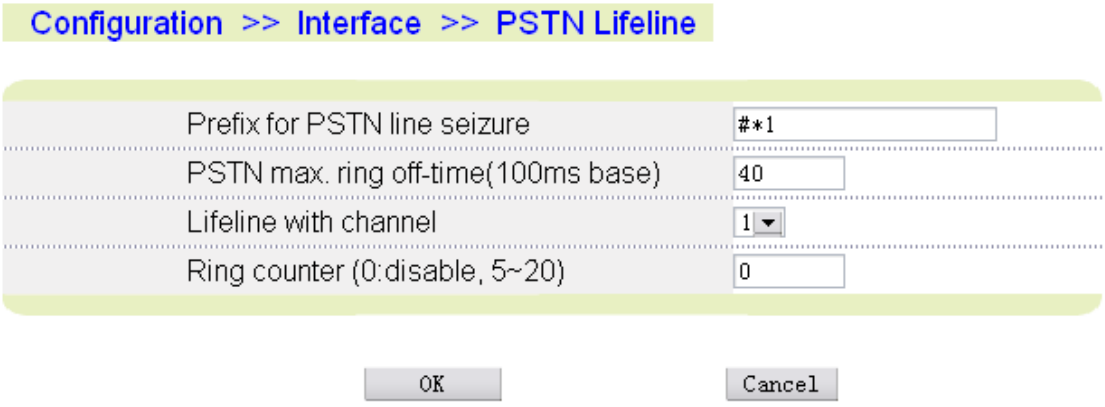


Figure 3-31 Analog (FXS/FXO) Configuration

Table 3-13 Description of PSTN Lifeline Items

Field	Description
Prefix for PSTN line seizure	Default: “#*1”.The function code to use PSTN Lifeline.
PSTN max. ring off-time	30..70. 100ms base. Default is 40.
Lifeline with channel	Default port for PSTN service when as callee. Default is port 1.
Ring counter	0: disable. Only ring on default port when PSTN call coming. 5..20 valid value. Ring for specify times on current spare port before turn to the next. The first one is the default port by setting.

If the configuration is finished, click “OK” button bellow the web page to confirm

changes or click “Cancel” button to deny it.

3.5.3 System Configuration

Table 3-14 The System Configuration Menu Items

Menu	Description
General	Configuration of the “Device name” and so on.
System Time	Simple Network Time Protocol (SNTP) client configuration.
User Management	Configuration of the three build-in users in VENUS 29xx: Administrator, operator and monitor.
UI Management	Configuration of build-in WEB and Telnet server
Provision	Configuration of the Provision
SNMP	Configuration of the Simple Network Management Protocol(SNMP) agent
TR069	Configuration of CPE WAN Management Protocol (CWMP or TR069) agent
FM Threshold	Fault alarm threshold configuration of CPU, DSP and RAM usage.
Syslog	Configuration of the capture and upload of the spy log messages.

3.5.3.1 General Configuration

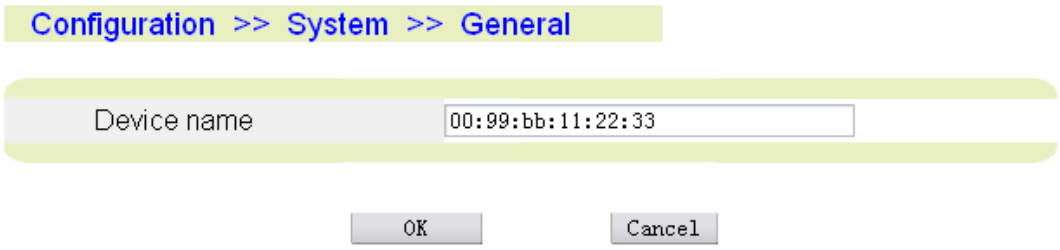


Figure 3-32 System General Configuration

Table 3-15 Description of System General Items

Field	Description
Device name	The device name (may be domain name) used by NMS. Default is the MAC of the device.

If the configuration is finished, click “OK” button bellow the web page to confirm

changes or click “Cancel” button to deny it.

3.5.3.2 System Time Configuration

Simple Network Time Protocol (SNTP) client functionality generates requests and reacts to the responses using the NTP version 3 protocol definitions (according to RFC 1305). Through these requests and responses, the NTP client is able to synchronize the system time to a time source within the network. By synchronizing time to a network time source, traffic handling, maintenance, and debugging actions become simplified for the network administrator.

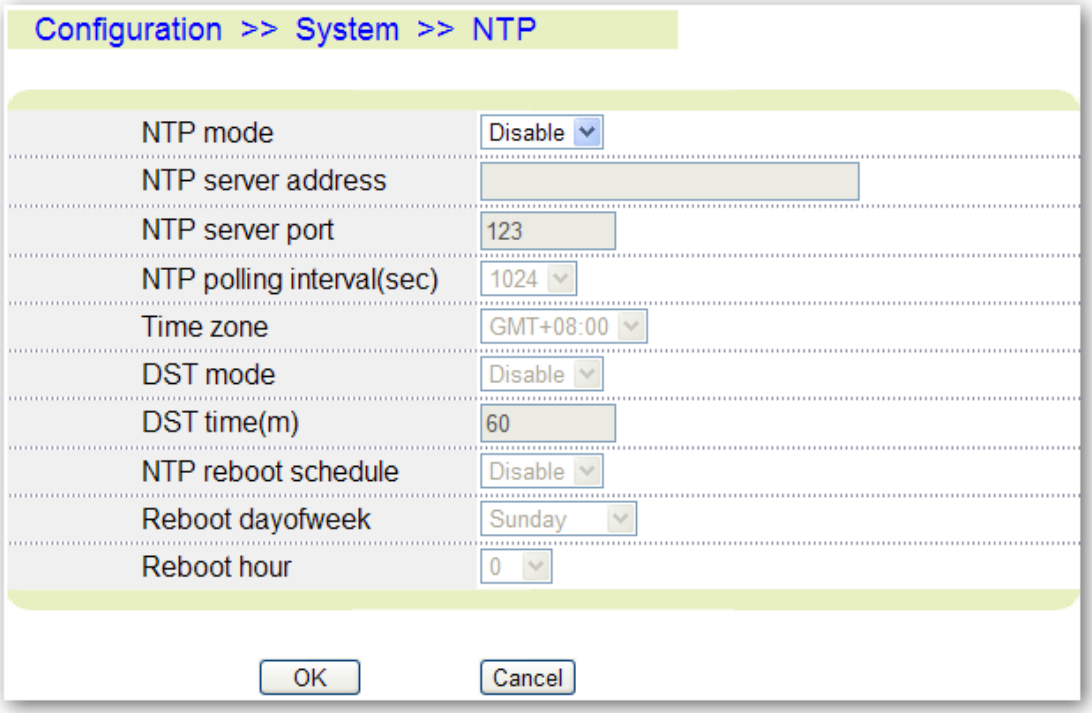


Figure 3-33 NTP Configuration

Table 3-16 Description of NTP Items

Field	Description
NTP Mode	1. Disable(default) 2. Enable
NTP server address	IP address (in dotted format notation) or domain name of the NTP server. Not assigned will disable the NTP function.
NTP server port	NTP service port number of NTP server. Default is 123.
NTP polling interval	Defines the time interval (in seconds) the NTP client requests for a time update. 64~8192. Default is 1024.

Field	Description
Time zone	Defines the GMT time zone (+13 to -12).
DST mode	Enable or Disable the Daylight saving time
DST time(m)	Minutes to be added to the current system time.
NTP reboot schedule	1. Disable(default) 2. Enable
Reboot dayofweek	Selection of reboot day of week when reboot schedule is enabled
Reboot hour	Configuration of NTP reboot hour in 24 hours format.

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.3.3 User Management Configuration

There are three build-in users in VENUS 29xx: Administrator, Operator and Monitor. User could change the name and password.

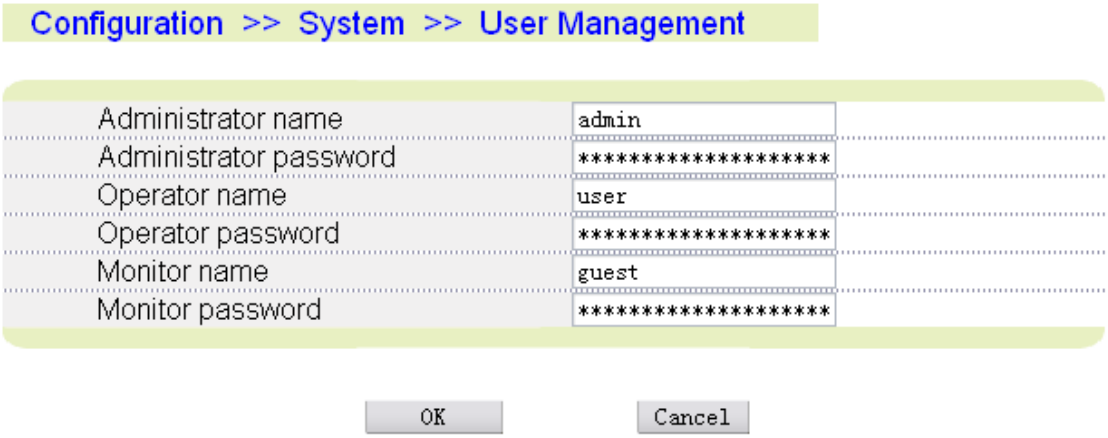


Figure 3-34 User Management Configuration

Table 3-17 Description of User Management Items

Field	Description
Administrator name	The administrator name and default is “admin”.
Administrator password	The administrator password and default is “admin”.

Field	Description
Operator name	The Operator name and default is “user”.
Operator password	The Operator password and default is NULL.
Monitor name	The Monitor name and default is “guest”.
Monitor password	The Monitor password and default is NULL.

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.3.4 UI Management Configuration

Venus 29xx Series has one build-in Telnet server and one build-in WEB server for remote management.

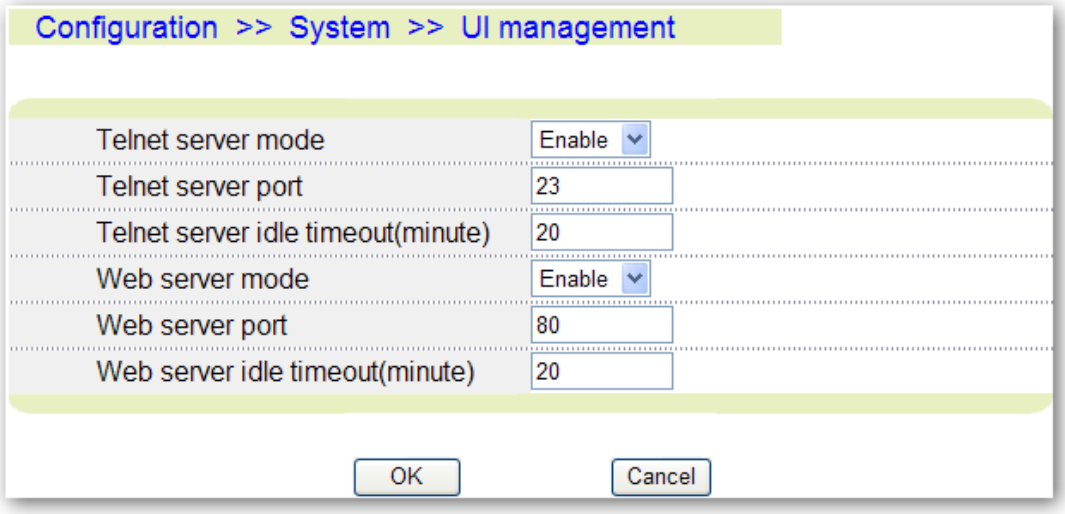


Figure 3-35 UI Management Configuration

Table 3-18 Description of UI Management Items

Field	Description
Telnet server mode	Enables or disables the build-in Telnet server. Telnet server is disabled by default in Ethernet interface 1. 1.Disable(Default) 2.Enable
Telnet server port	Telnet service port number of TCP. Default is 23

Field	Description
Telnet server idle timeout	Sets the timeout for disconnection of an idle web session (in minutes). When set to zero, idle sessions are not disconnected. Default is 64.
Web server mode	Enables or disables the build-in Web server. Web server is enabled by default in Ethernet interface 1. 1.Disable 2.Enable (Unsecured)(Default)
Web server port	Web service port number of TCP. Default is 80.
Web Server Idle Timeout	Sets the timeout for disconnection of an idle web session (in minutes). When set to zero, idle sessions are not disconnected. Default is 20.

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.3.5 Provision

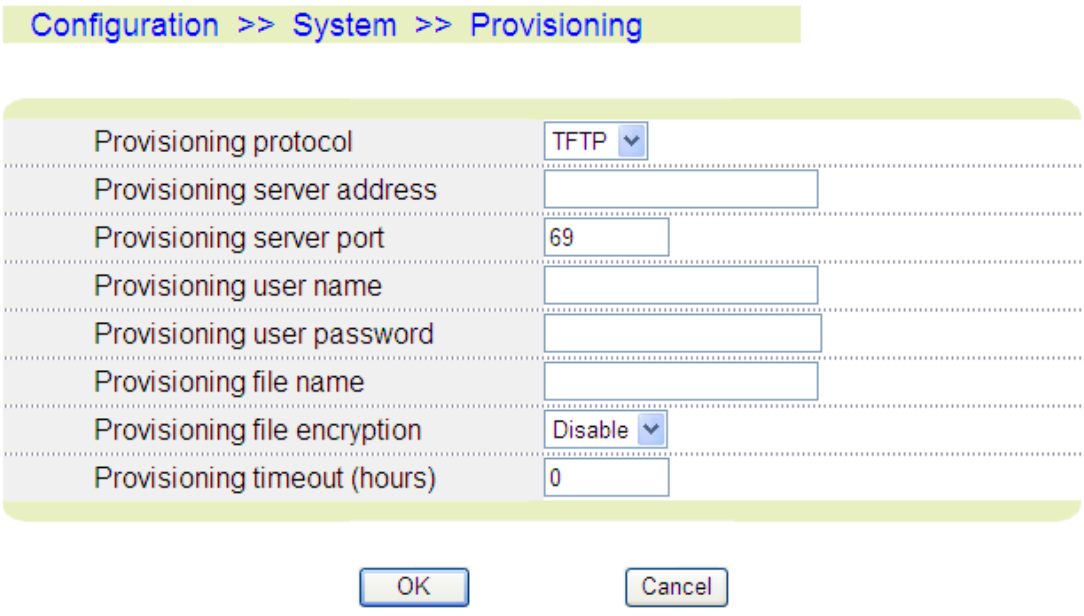


Figure 3-36 Provisioning Configuration

Table 3-19 Description of Provisioning Items

Field	Description

Field	Description
Provisioning protocol	The protocol which is used for downloading provisioning file. Default: TFTP, FTP, HTTP
Provisioning server address	The provisioning server IP address
Provisioning server port	The provisioning server port
Provision user name	The user name is used for authentication
Provisioning user password	The password is used for authentication
Provisioning file name	The name of provisioning file
Provisioning file encryption	Default :disable Enable, use encryption file. The encryption algorithm is PC1.
Provisioning timeout	The timeout of twice provisioning file acquire. When set to zero, just acquire provisioning file once in VENUS 29xx restart.

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.3.6 SNMP

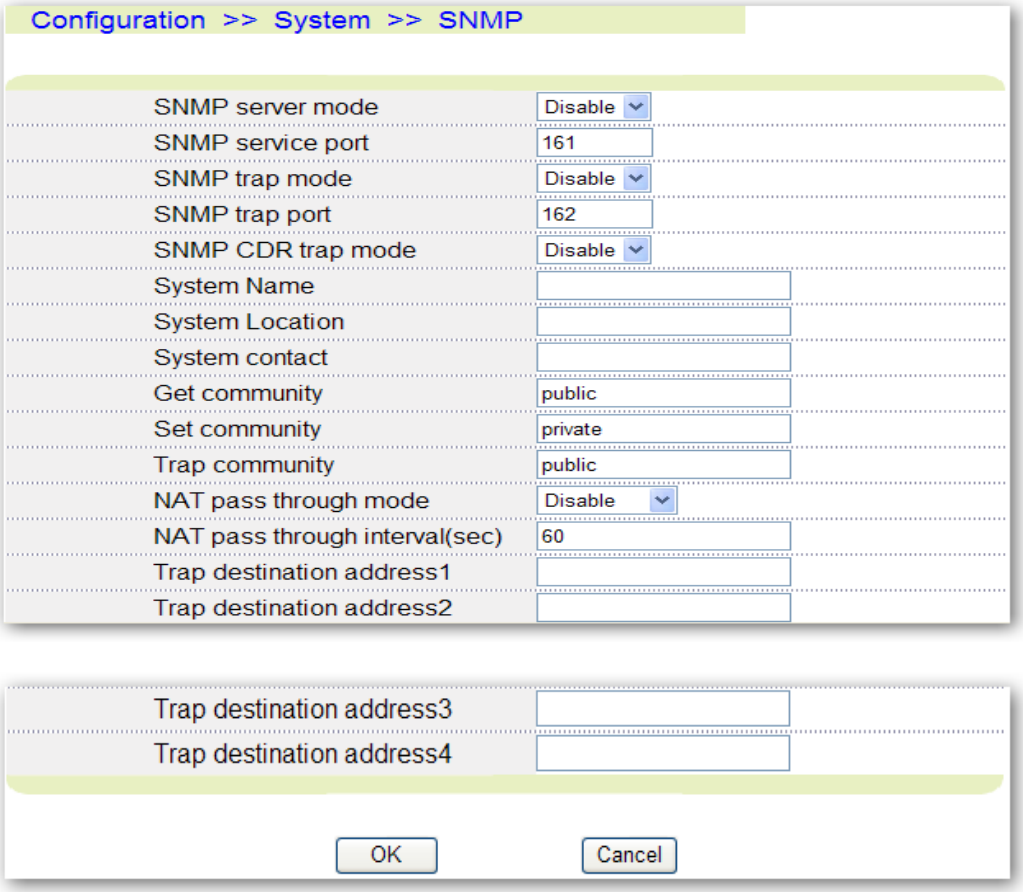


Figure 3-37 SNMP Configuration

Table 3-20 Description of SNMP Items

Field	Description
SNMP server mode	Default: disable Enable: start up SNMP agent in VENUS 29xx
SNMP service port	The port of SNMP agent ,default is 161
SNMP trap mode	Default: disable Enable: enable send trap function.
SNMP trap port	The port for sending trap to, default is 162
SNMP CDR trap mode	Default: disable Enable: enable send CDR trap function.
System name	The system name
System location	The system location
System contact	The contact information
Get community	The get community and default is public
Set community	The set community and default is private
Trap community	The trap community and default is public
NAT pass through mode	Default:disable Enable: enable this option ,just when VENUS 29xx is in private network
NAT pass through interval(sec)	The keep alive time for SNMP NAT
Trap destination address 1-4	The destination IP address for sending trap. VENUS 29xx can support four destinations.

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.3.7 TR069

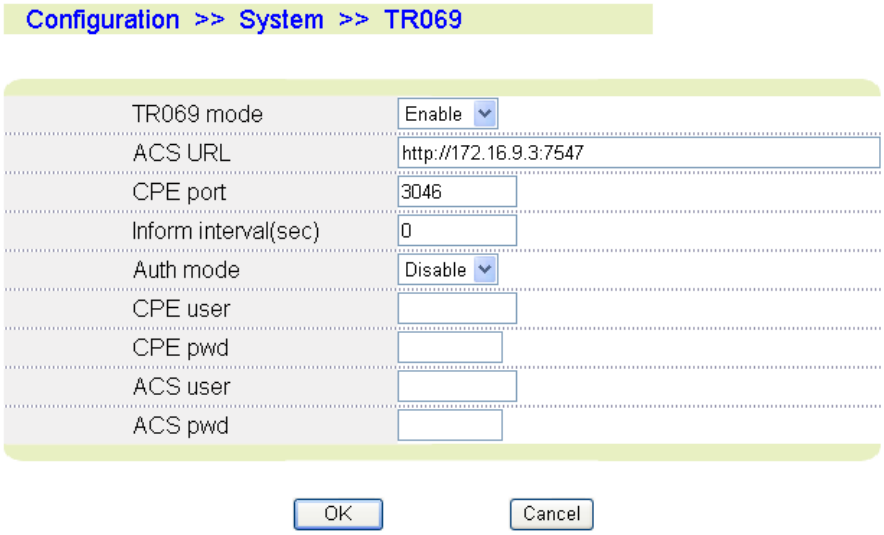


Figure 3-38 TR069 Configuration

Table 3-21 Description of TR069 Items

Field	Description
TR069 mode	Default: disable Enable: start up SNMP agent in VENUS 29xx
ACS URL	Auto-Configuration Server. This is a component in the broadband network responsible for auto-configuration of the CPE for advanced services.
CPE port	The port of CPE
Inform interval(sec)	The inform time interval in seconds
Auth mode	1.Disable(Default) 2.Basic 3.Digest
CPE user	CPE user name
CPE pwd	CPE password
ACS user	ACS user name
ACS pwd	ACS password

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.3.8 FM Threshold

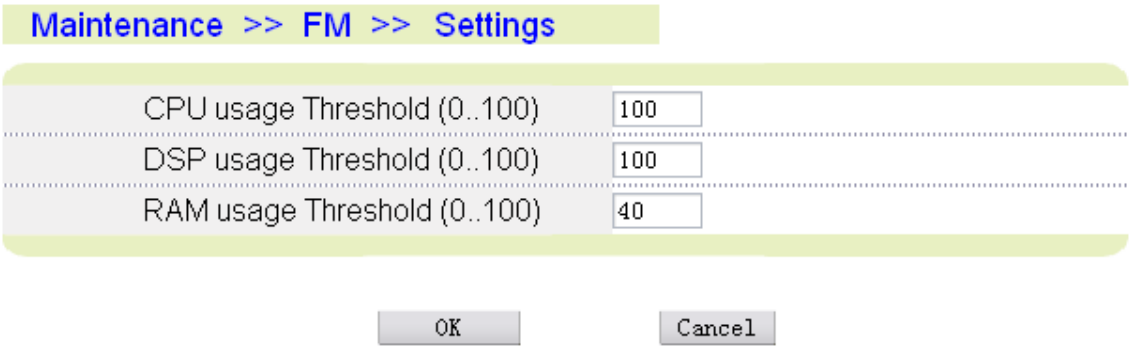


Figure 3-39 FM Configuration

Table 3-22 Description of FM Threshold Items

Field	Description
CPU usage Threshold (0..100)	If current CPU usage exceeds the threshold create an alarm message. Default: 100(%)
DSP usage Threshold (0..100)	If current DSP usage exceeds the threshold create an alarm message. Default: 100(%)
RAM usage Threshold (0..100)	If current RAM usage exceeds the threshold create an alarm message. Default: 100(%)

3.5.3.9 Syslog

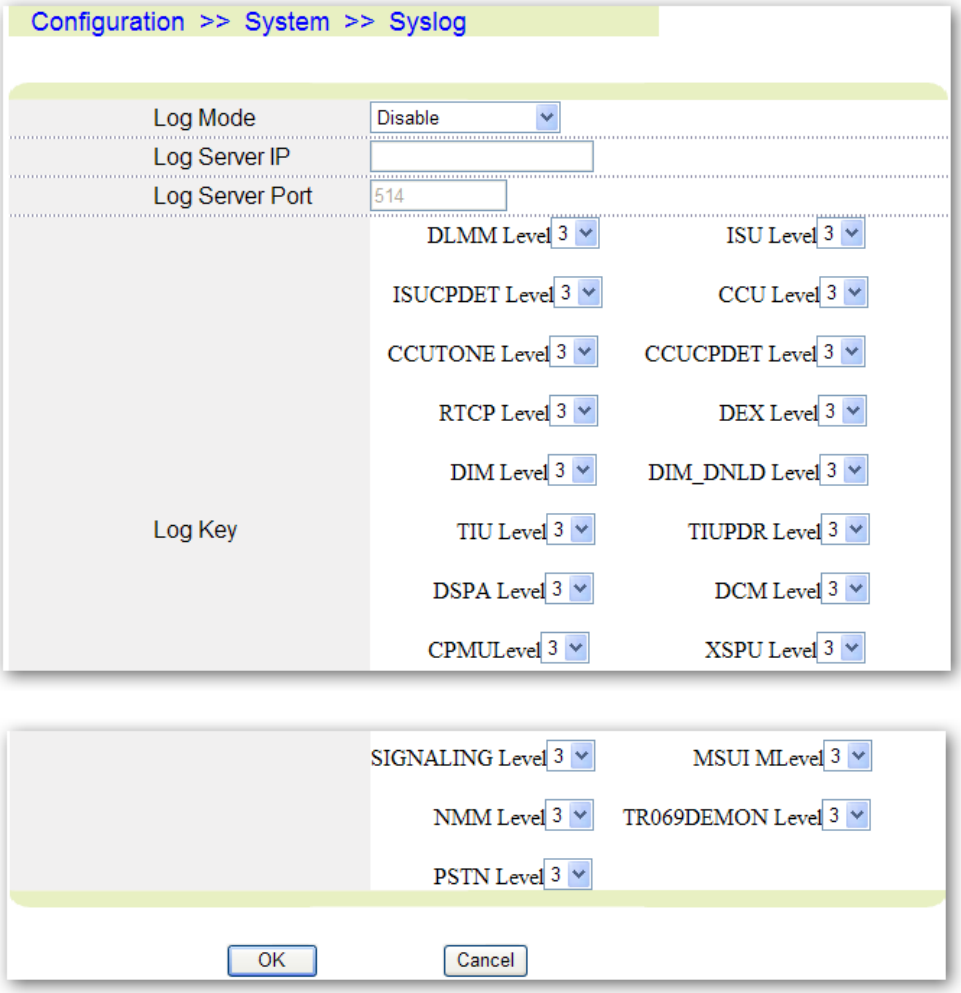


Figure 3-40 Syslog Configuration

Table 3-23 Description of Syslog Items

Field	Description
Log Mode	1. Disable(Default) 2. Remote Server 3. Local File System
Log Server IP	IP address of the Log Server
Log Server Port	Port number of the Log Server. Default is 514.

3.5.4 SIP Configuration

3.5.4.1 SIP General Configuration

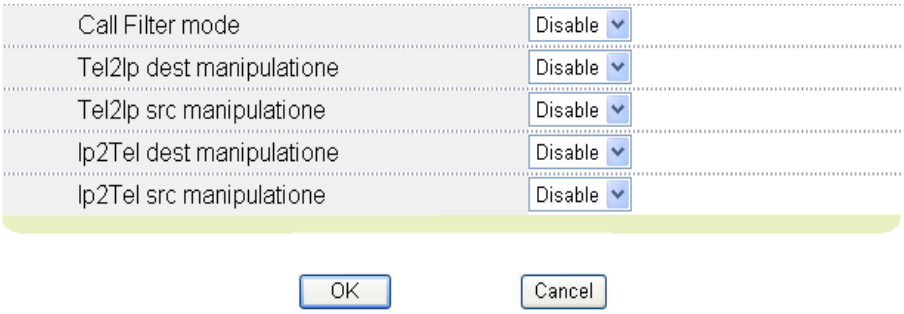


Figure 3-41 SIP General Configuration

Table 3-24 Description of SIP General Configuration Items

Field	Description
FQDN mode	Enable/Disable FQDN function. Default is Disable
FQDN name	When FQDN mode is Enable, input FQDN name. Used for specify the contact URL to a special address.
Trust proxy	1. Disable (Default) 2. Enable, Check the remote IP and port of the message has received whether it matches the configured Proxy address, Registrar address or Outbound Proxy address . If it isn't match any one of them ,this message will be discard.This function can be use to avoid packets from unknown address.
T1 time	An estimate of the round-trip time (RTT) for SIP protocol, default value is set to 500 ms.
Password with md5	1. Disable (Default) 2. Enable
Reg auth always	1. Disable (Default) 2. Enable
Use port in URI	1. Disable (Default), Use the default port 5060 for the request URL (Effect the request line and the destination). 2. Enable, Use the port of the realm configuration for the request URL (Effect the request line and the destination).
Compact mode	1. Disable (Default) use long format header type 2. Enable use short format header type
Invite without SDP	1. Disable (Invite with SDP), Default 2. Enable
In band ring back tone	1. Disable (Default) 2. 183 with SDP 3. 180 with SDP
Invite timeout	Default is 60(0~65536). The default Invite timeout in seconds.

Session Expires Mode	1. Disable (Default) 2. Use UPDATE 3. Use Re-INVITE
Session Expires Time	3600 (Default). Determines the timeout (in seconds) for keeping a re-INVITE message alive within a SIP session. The SIP session is refreshed each time this timer expires. The SIP method used for session-timer updates is determined according to the Session Expires Mode.
Minimum Session Expires (10~100000)	120(Default). Defines the time (in seconds) that is used in the Min-SE header. This header defines the minimum time that the user agent supports for session refresh. The valid range is 10 to 100000.
PRACK mode	1. Disable (Default), Not support PRACK. 2. Supported, Support PRACK but not required. 3. Required, Require PRACK.
INFO Mode	1. Disable (Default). 2. Enable, UE can send DTMF signal through INFO message.
Subscription mode	1. Disable, Default 2. Enable; Register subscription.
Pound Symbol mode	1. Disable (Default), use %23 if need to report # in URI 2. Enable, use # in Invite URI
Ring mode	Ring mode in hunting group 1. Ascending any one (Default). Select the lowest available channel. 2. Ascending cyclic. Select the lowest available channel number in the hunt group. Select the next higher available channel number when no answer. 3. All in hunting group. Select all channels in hunting group and the first hanging down channel will be connected.
Current disconnect	1. Disable, Default 2. Enable; Note：This option just for future use
Routing table mode	Determines if the local Tel to IP routing table takes precedence over a Proxy for routing calls. 1. Disable (Default), Prefer to use Proxy to route calls. 2. Prefer, The gateway checks the 'Dest Phone Prefix' and/or 'Source Phone Prefix' field in the 'Tel to IP Routing' table for a match with the outgoing call. Only if a match is not found, a Proxy is used

	3. Fallback, If use Proxy to route calls fail the gateway checks the 'Dest Phone Prefix' and/or 'Source Phone Prefix' field in the 'Tel to IP Routing' table for a match with the outgoing cal
Use "user=phone" in SIP URL	1. Disable, Default 2. Enable
Tel Invitation URL mode	1. Disable (Default), Prefer to use SIP format number in to URL of the Invite request. 2. Enable, Prefer to use Tel format number in to URL of the Invite request.
Tel Register URL mode	1. Disable (Default), Prefer to use SIP format number in the Register request. 2. Enable, Prefer to use Tel format number in the Register request.
Register Fail No Resend	1. Disable (Default), Prefer to resend the Register request when register fail. 2. Enable, Prefer to not resend the Register request when register fail.
UNRegister Clean Call	1. Disable (Default), Prefer to not clean the call of the user when it is unregister. 2. Enable, Prefer to clean all the call of the user when it is unregister.
Service Route mode	1. Disable (Default), Prefer to not to use the service route. 2. Enable, Prefer to use the service route.
Early Media mode	1. Disable, Default 2. Sendrecv 3. Sendonly 4. Recvonly 5. Inactive 6. Gated 7. Supported
Request Disposition mode	1. Disable, Default 2. No-fork 3. Parallel 4. Sequential
Call Filter mode	1. Disable, Default 2. Enable
Tel2Ip dest manipulatione	1. Disable, Default 2. Enable

Tel2Ip src manipulatione	1. Disable, Default 2. Enable
Ip2Tel dest manipulatione	1. Disable, Default 2. Enable
Ip2Tel src manipulatione	1. Disable, Default 2. Enable

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.4.2 SIP Realm Configuration

User can use different realm to get different Registrar and Proxy parameters for each analog line. Venus 29xx Series allow user to input 8 (REALM_NUMBER) groups of parameter for selection and the index is 0 to 7. The realm configuration parameter is as below:

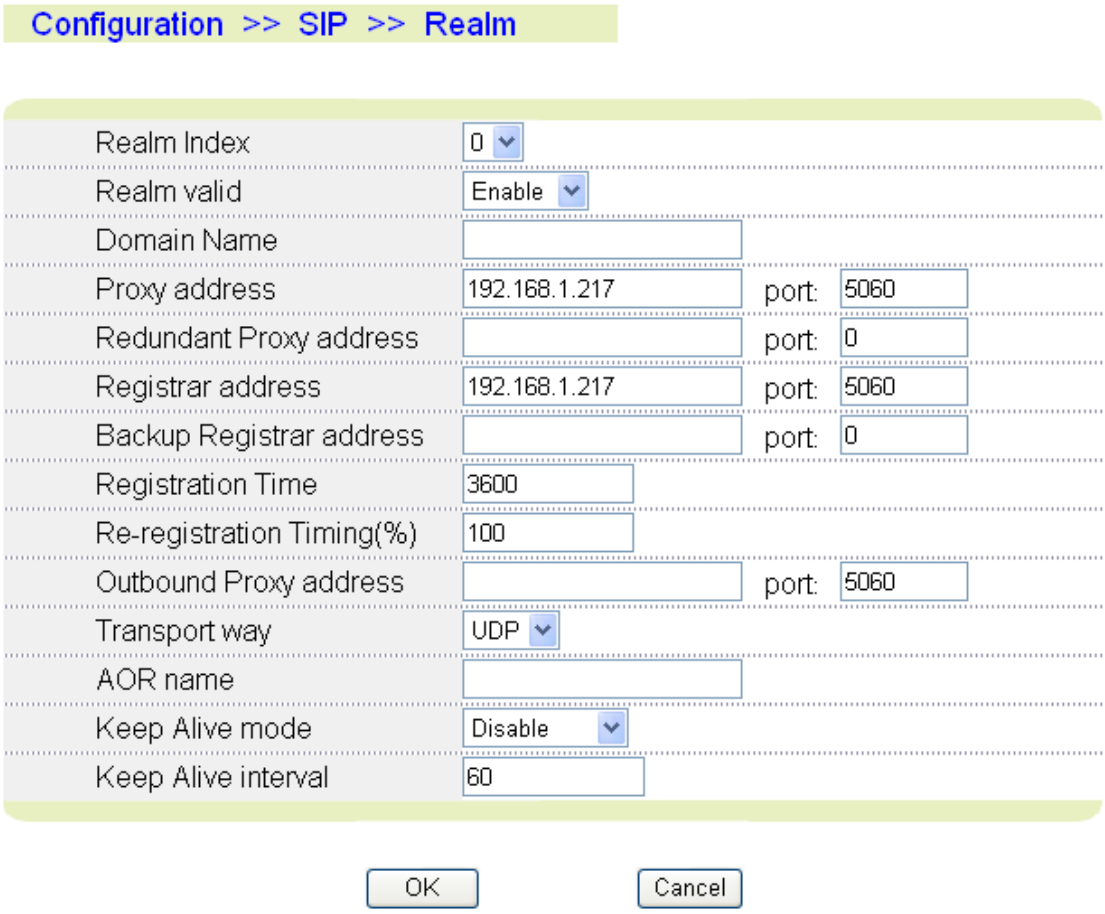


Figure 3-42 SIP Realm Configuration

Table 3-25 Description of SIP Realm Configuration Items

Field	Description
Realm Index	Quick Start parameter The assigned realm index (0~7) Where, 0 to 7 means the used realm.
Realm valid	1. Disable (Default) 2. Enable
Domain Name	Quick Start parameter Defines the Domain Name of specified realm. If specified, the Domain Name is used as Request-URI in REGISTER, INVITE and other SIP messages. If not specified, the Registrar/Proxy address is used instead.
Proxy address	Quick Start parameter IP address or domain name of the primary Proxy server you are using. If you enable Proxy Redundancy, the gateway can work with tow Proxy servers. If there is no response from the primary Proxy, the gateway tries to communicate with the redundant Proxy.
Proxy port	5060(Default) Quick Start parameter Primary proxy service port number.
Redundant Proxy address	NULL (Default) IP addresses or domain name of the redundant Proxy you are using.
Redundant Proxy Port	0(Default) Service port number of the redundant Proxy you are using.
Registrar address	Quick Start parameter IP address or domain name of the primary Registrar server you are using. If this parameter is not specified and analog line need do REGISTER, the proxy server is used.
Registrar Port	5060(Default) Quick Start parameter Primary registrar service port number.
Backup Registrar address	NULL(Default) IP address or domain name of the backup Registrar server you are using. Unlike redundant proxy server, VENUS 29xx will send REGISTER message even if the primary Registrar works well.
Backup Registrar	0(Default)

Port	Backup registrar service port number.
Registration Time	3600(Default) Defines the time (in seconds) for which registration to a Proxy server is valid. The value is used in the header 'Expires'. If Registrar server return new expires parameter in response, the gateway resumes registration according to the new parameter.
Re-registration Timing (%)	100(Default) Defines the re-registration timing (in percentage). The timing is a percentage of the re-register timing set by the Registration server. The valid range is 50 to 100. The default value is 100. For example: If 70 (%) is used and Registration Expires time is 3600, the gateway resends its registration request after 3600 x 70% = 2520 sec.
Outbound Proxy address	Quick Start parameter IP address or domain name of the outbound proxy server.
Outbound Proxy Port	Quick Start parameter Outbound proxy service port number.
Transport way	Quick Start parameter Transport protocol and valid options include:
AOR name	If AOR name is specified, REGISTER message will use this parameter in From/To URI. Otherwise, it will use the same value of Request URI.
Keep Alive mode	1. Disable (default). 2. Enable Keep alive with Proxy using OPTIONS. 3. Enable Keep alive with Proxy using REGISTER. Any response from the Proxy, either success (200 OK) or failure (4xx response) is considered as if the Proxy is correctly communicating. Note 1: This parameter must be set to 2(OPTIONS) when Proxy redundancy is used. Note 2: When the active proxy does not respond to INVITE messages sent by the gateway, the proxy is marked as inactive. The behavior is similar to a Keep-Alive (OPTIONS or REGISTER)
Keep Alive interval	60(Default) Defines the Proxy keep-alive time interval (in seconds) between Keep-Alive messages.

If the configuration is finished, click "OK" button bellow the web page to confirm changes or click "Cancel" button to deny it.

3.5.4.3 SIP Group Configuration

The analog line could be managed in hunting group and we assume the maximum group number is 8 and valid index is 0 to 7. The hunting group is a logical UA and may contain multiple physical analog ports.

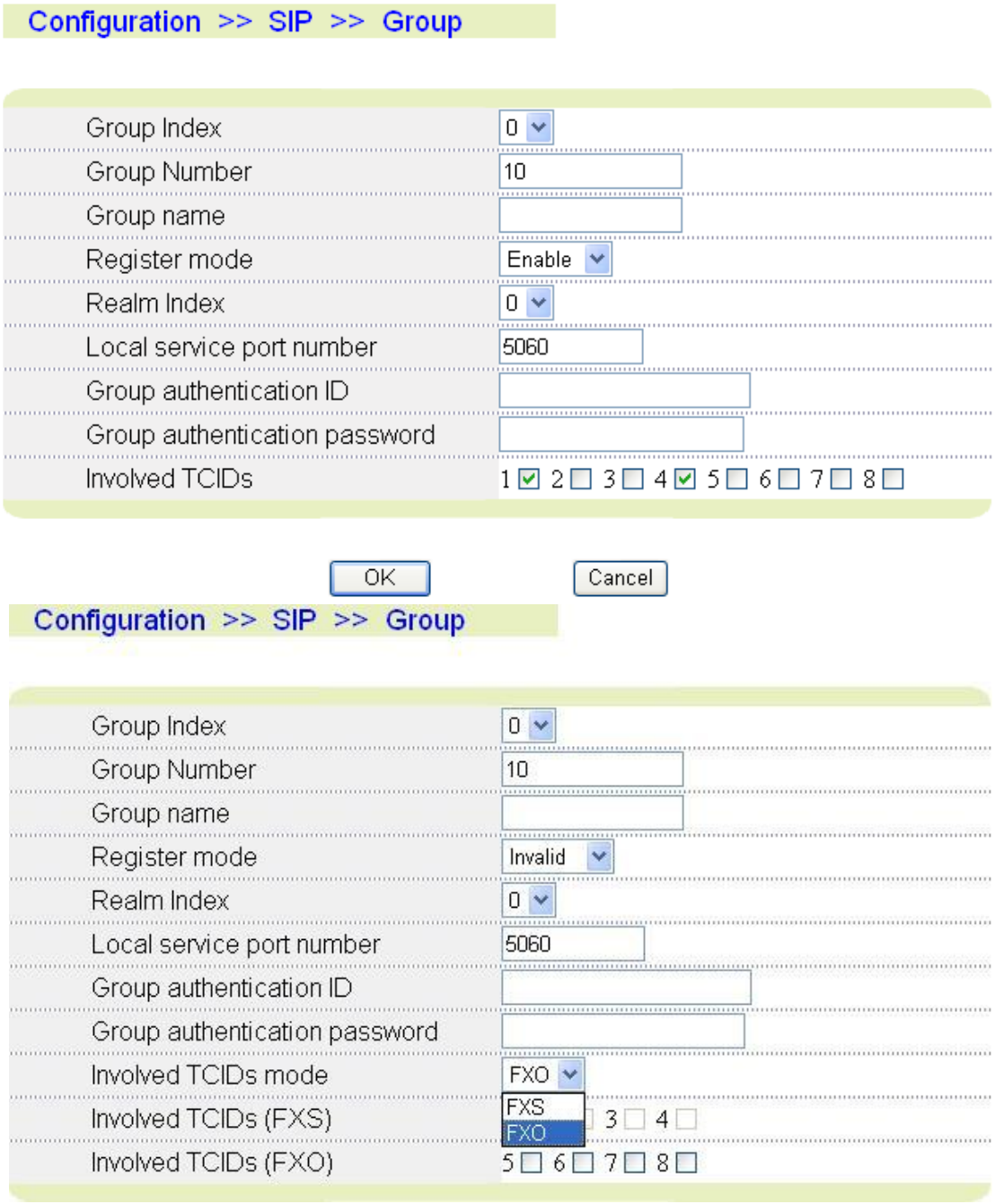


Figure 3-43 SIP Group Configuration for 8S and 4S4O Model

Table 3-26 Description of SIP Group Configuration Items

Field	Description
Group Index	Quick Start parameter The assigned realm index (0~7) Where, 0 to 7 means the used realm.
Group Number	Quick Start parameter The register number and allow use the same number for different port. The ports used the same number is taken as group binding number.
Group name	The display name for specified group.
Register mode	Quick Start parameter 1. Invalid 2. Disable 3. Enable
Realm Index	Quick Start parameter The assigned realm index (0~7) Where, 0 to 7 means the used realm.
Local service port number	5060(Default)
Group authentication ID	Quick Start parameter The authentication user ID.
Group authentication password	Quick Start parameter The authentication user password.
Involved TCID Mode (4S4O Model Only)	FXS FXO
Involved TCIDs	Set the ports of the group.

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.4.4 SIP Phone Routing Table Configuration

The Phone Routing Table (build-in phone book) is used to route incoming telephone calls to IP addresses. This routing table associates a called / calling telephone number’s prefixes with a destination IP address or with an FQDN (Fully Qualified Domain Name).

When Proxy server is not used in VENUS 29xx, a call is routed through the Phone

routing table. The called and calling numbers are compared to the list of prefixes on the Phone Routing Table (up to 50 prefixes can be configured). Calls that match these prefixes are sent to the corresponding IP address. If the number dialed does not match these prefixes, the call is not made.

When using a Proxy server, you do not need to configure the Phone Routing Table. However, if you want to use fallback routing when communication with Proxy servers is lost, or to obtain different SIP URI host names (per called number), you need to configure the Phone Routing Table.

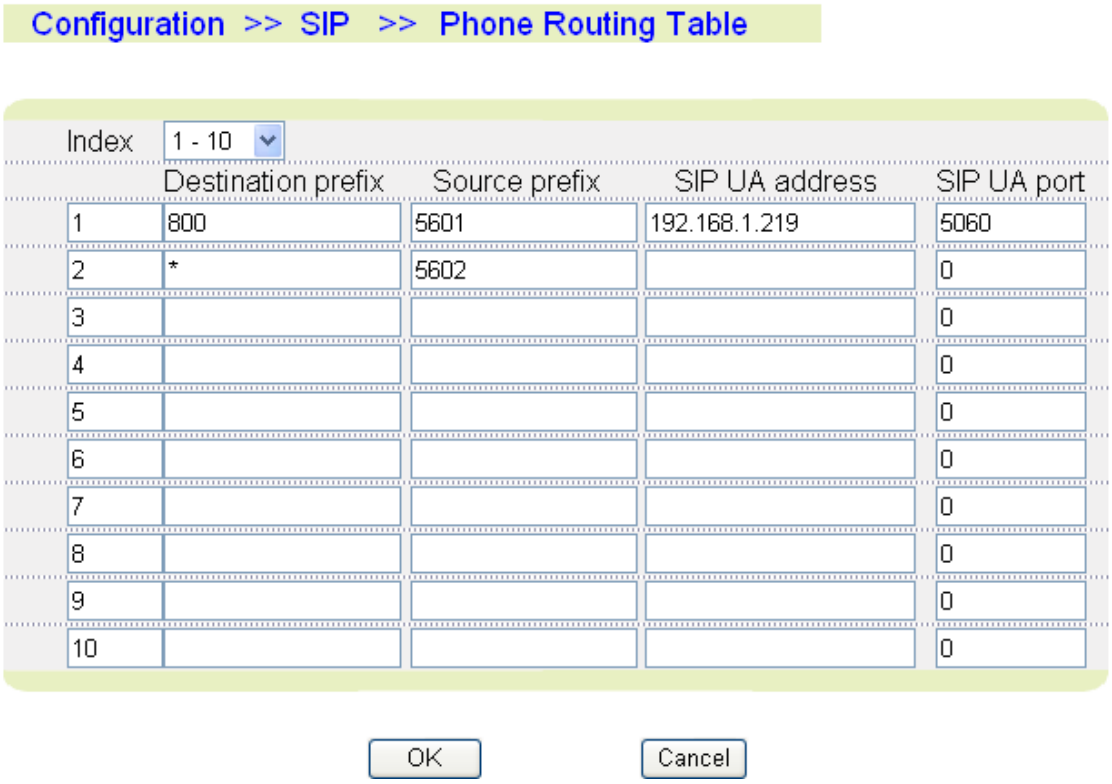


Figure 3-44 SIP Phone Routing Table Configuration

Table 3-27 Description of SIP Phone Routing Table Configuration Items

Field	Description
Destination prefix	Each entry in the Destination phone number prefix fields represents a called telephone number prefix. The prefix can be 1 to 32 digits long. An asterisk (*) represents all numbers.
Source prefix	Each entry in the Source phone number prefix fields represents a calling telephone number prefix. The prefix can be 1 to 32 digits long. An asterisk (*) represents all numbers.
SIP UA address	In each of the SIP Address fields, enter the IP address that is assigned to these prefixes. Domain names can be

	used instead of IP addresses. To discard outgoing IP calls, enter 0.0.0.0 or nothing in this field.
SIP UA port	5060(Default) Remote SIP UA service port number.

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.4.5 SIP Digit Manipulation Configuration

The digit manipulation is a more specific digit map used to control the number to be dialed out. The manipulation table is able to specify the number of digits to be removed or added.

The digit manipulation rules are executed in the following order:

- Number of stripped digits.

- Number of digits to leave(keep).

- Prefix / suffix to add.

First two parts (**Dst prefix** and **Scr prefix**) are monitoring fields to see if the **dialed number(To which number)** or the **dialing number(From which number)** matching the rule.

- **Dst prefix:** Checking **the number which the user just dialed** and see if it matches the prefix number configured in this field.

For example, if current Dst prefix has an entry of 888, then numbers such as 88811, 888535, 88823582 would match this rule.

- **Src prefix:** Checking if the calling user port/extension number matches the prefix number configured in this field.

For example, if current port extension numbers of 2908 are 5501, 5502, 5503, 5504, 6601, 6602, 6603, 6604 and Src prefix is configured as 66.

Then only the last four ports, 6601 to 6604, would be affected by this particular rule.

User could also input " ***** " to apply this rule to any port in 2908.

Next three parts (**Strip Digit, Digit Leave,** and **Prefix(Suffix)**) are modification to the dialed number, remove or add digits as required.

- **Strip Digit:** The number of digits to be removed from the number which the user just dialed.

Usage: Enter the number of digits to remove the digits **from beginning of the number**.

Enter the number of digits in brackets "( )" to remove the digits **from end of the number**.

For example, if user dials 12345678, and configure Strip Digit to be "2(3)", then Venus will

remove 2 digits in the front and 3 digits at the end.

Resulting final output would be 345 only, because digits were removed.

- **Digit Leave:** The **number of digits to keep** from the original number or after digit stripped.

For example, if the final number dialed by the user is 345 or the final number after digit stripped is 345, and Digit Leave is configured as 2.

The final output would be only 45 because only the last 2 digits will be kept.

- **Prefix (Suffix): Prefix**: The number to be **added in front** of the dialed or final output number.

Suffix: The number to be **added at the end** of the dialed or final output number.

For example, if the final dialed number or original number is 45 and Prefix(Suffix) is configured as 888(9999).

The final result would be 88845999.

Here is an example of all the rules combined together and resulting a final number to dial out.

- Settings: Dst prefix: 2; Src prefix: 70; Strip Digit:2(1); Digit Leave: 2; Prefix(Suffix): 55(1234).

- User "7001" dial "29080123";

"29080123" --> AfterStrip Digit: 08012; -->After Digit Leave: 12; -->Pre(Suff)Fix:55121234;

-->The final result is "55121234".

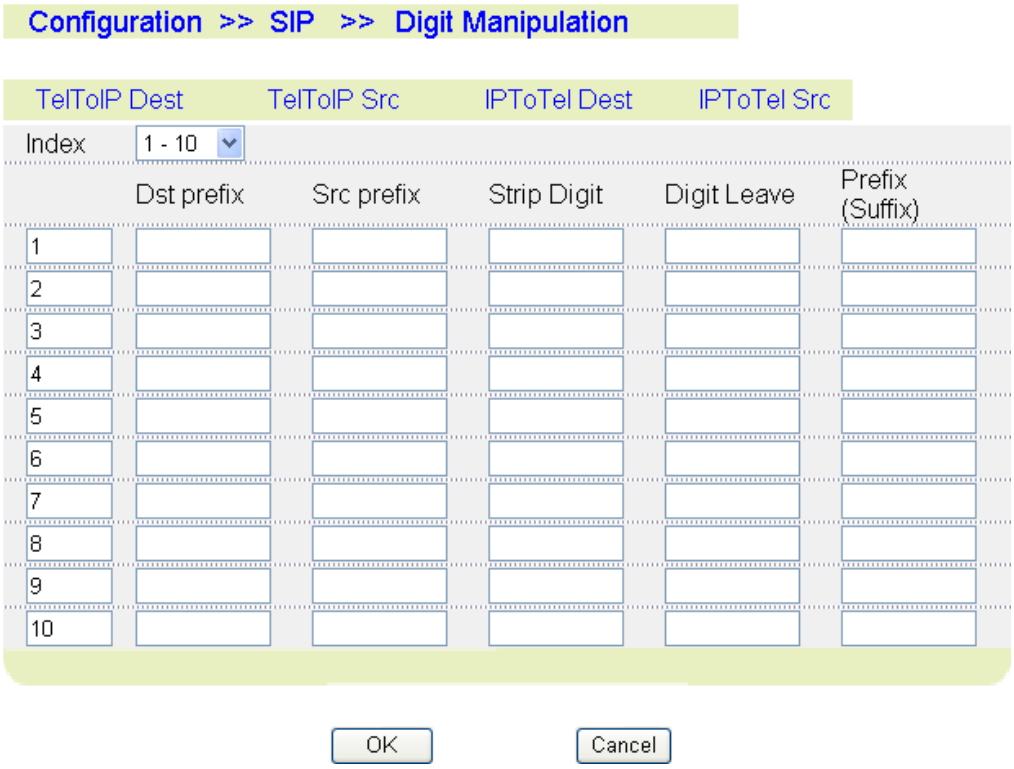


Figure 3-45 SIP Digit Manipulation Configuration

Table 3-28 Description of Digit Manipulation Items

Field	Description
Dst prefix	Each entry in the Destination phone number prefix fields represents a called telephone number prefix. The prefix can be 1 to 32 digits long. An asterisk (*) represents all numbers.
Src prefix	Each entry in the Source phone number prefix fields represents a calling telephone number prefix. The prefix can be 1 to 32 digits long. An asterisk (*) represents all numbers.
Strip Digit	Number of digits to be removed from the left of the telephone number prefix, or enter the number of digits (in brackets) to remove from the right of the telephone number prefix.
Digit Leave	Number of digits to be left from the right
Prefix(Suffix)	Enter the number/string to be added to the front of the phone number, or enter the number in brackets to add the number to the end of the phone number.

3.5.5 Advanced Configuration

3.5.5.1 RTP Management Configuration

RTP parameters are used to assign the RTP base port number, RTCP mode, RTP filter working mode and CODEC associated parameters.

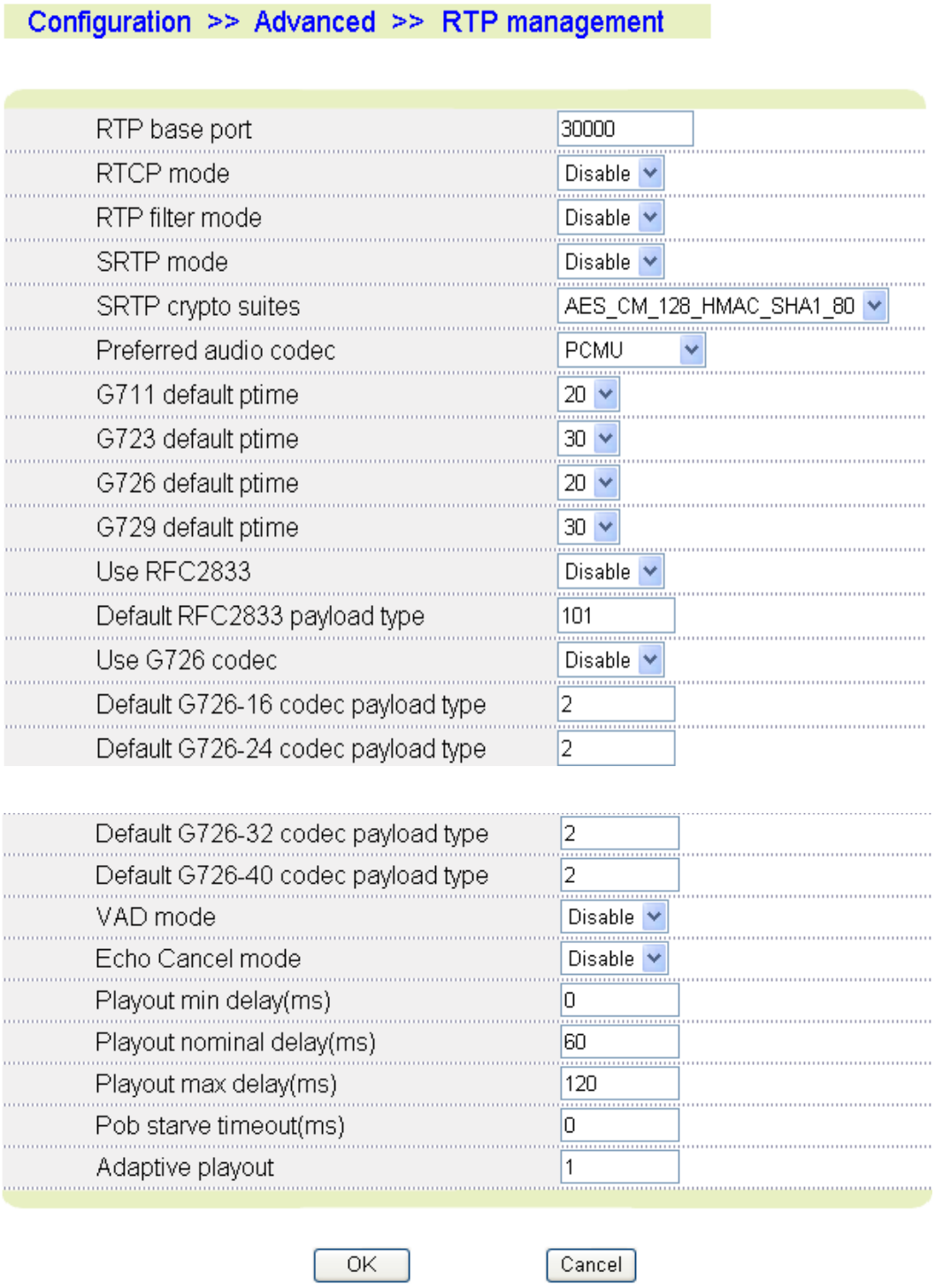


Figure 3-46 RTP Management Configuration

Table 3-29 Description of RTP Management Configuration Items

Field	Description
RTP base port	30000(Default) The base port number and total 32 numbers are reserved for specific group.
RTCP mode	Enable/Disable (Default) RTCP.

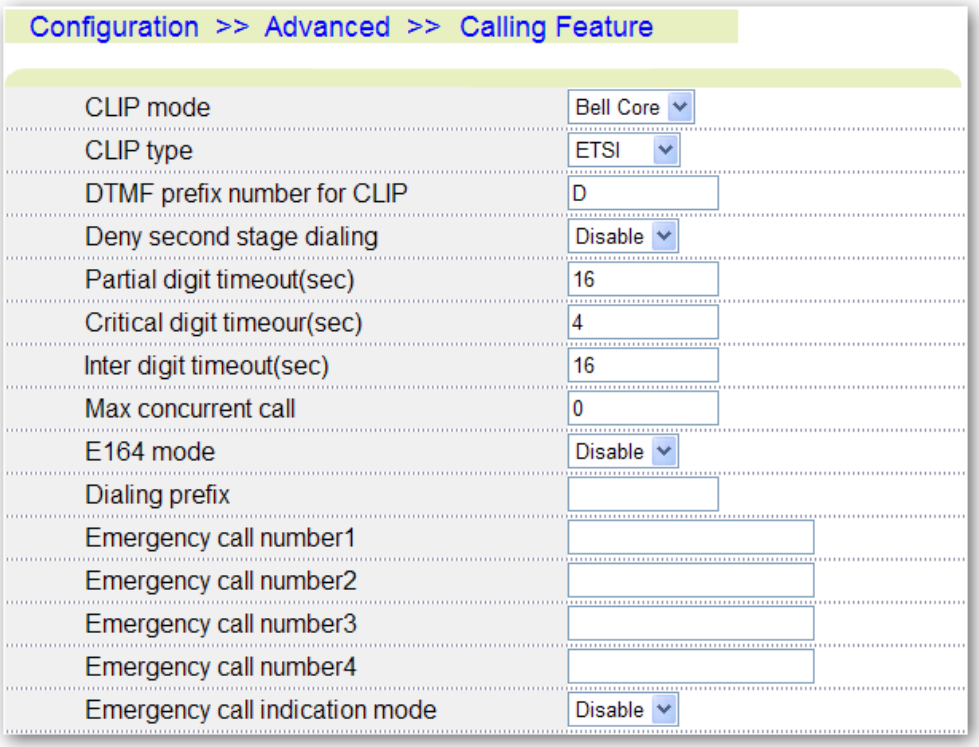
RTP filter mode	1. Disable(Default) 2. Enable, can filter RTP packages when RTP source not match.
SRTP mode	1. Disable (Default) 2. Enable
SRTP crypto suites	1.AES_CM_128_HMAC_SHA1_80 (Default) 2. F8_128_HMAC_SHA1_80
Preferred audio codec	1.PCMU 2.PCMA 3.G.729A 4.G.729AB 5.G.723 (5.3K) 6.G.723 (6.3K) 7.G.726-16 8.G.726-24 9.G.726-32 10.G.726-40
G711 default ptime	20(Default) Valid value is 10, 20 and 30 in ms.
G723 default ptime	30(Default) Valid value is 30 and 60 in ms.
G726 default ptime	20(Default) Valid value is 10, 20 and 30 in ms.
G729 default ptime	20(Default) Valid value is 10, 20, 30, 40, 50 and 60 in ms.
Use RFC2833	RFC2833 enable or disable (Default).
Default RFC2833 payload type	101(Default) The default payload type of RFC2833. The real time payload type may be negotiated with SDP.
Use G726 codec	1.Disable (Default) 2.Enable (Use G726 codec)
Default G726-16 codec payload type	2 (Default) The default payload type of G726-16.
Default G726-24 codec payload type	2 (Default) The default payload type of G726-24.
Default G726-32 codec payload	2 (Default) The default payload type of G726-32.

type	
Default G726-40 codec payload type	2 (Default) The default payload type of G726-40.
VAD mode	Enable/Disable (Default) VAD function
Echo Cancel mode	Enable/Disable (Default) EC function
Playout min delay	Playout minimum delay (ms)
Playout nominal delay	Playout nominal delay (ms)
Playout max delay	Playout maximum delay (ms)
Pob starve timeout	Pob starve timeout (ms)
Adaptive playout	Adaptive playout

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.5.2 Calling Feature Configuration

These parameters are used to define the call feature for all protocol including SIP.



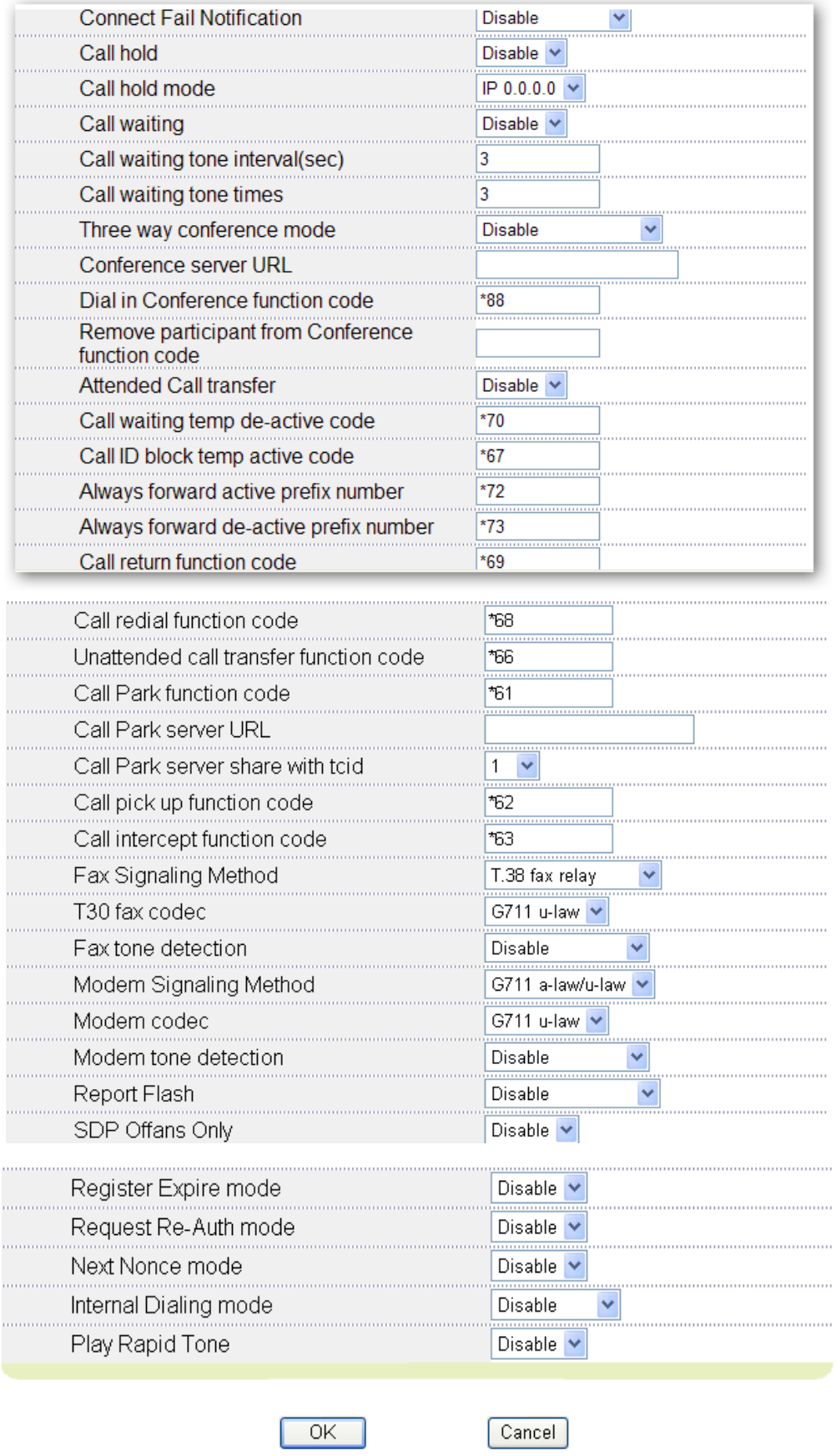


Figure 3-47 Calling Feature Configuration

Table 3-30 Description of Calling Feature Configuration Items

Field	Description
CLIP mode	Select from Bell Core, ETSI and DTMF for Calling Line Identification Presentation.
CLIP type	Select from NA, Japan, ETSI, YDN, BT and Canada for the type of Calling Line Identification Presentation.
DTMF prefix number for CLIP	D(Default)This is just for DTMF caller ID display.
Deny second stage dialing	1. Disable (Default) 2. Enable, Deny second stage dialing if needed.
Partial digit timeout	The time between partial digits
Critical digit timeout	The time between Critical digits
Inter digit timeout	The time between two digits input.
Max concurrent call	The max concurrent call is allowed in the device. 0, not limit;
E164 mode	Disable (Default)/ Enable (E164 function)
Dialing prefix	To append the prefix to the dial string. Used for all the users of the device to share a dialing prefix.
Emergency call number1	The emergency call number
Emergency call number2	The emergency call number
Emergency call number3	The emergency call number
Emergency call number4	The emergency call number
Emergency call indication mode	1.Disable (Doesn’t ring when caller hung up and release the call) 2. Enable (Ring when caller hung up and remain t the call)
Call hold	1. Disable (Default) 2. Enable (support music on hold, no checking)
Call hold mode	1. IP 0.0.0.0(Default) 2. SendOnly 3. Inactive
Call waiting	1. Disable (Default) 2. Enable
Call waiting tone interval	3 (Default value) The call waiting tone interval time in seconds.
Call waiting tone	Set times playing Call waiting tone. (Invalid for Current

times	Version)
Three Way conference mode	1. Disable (Default) 2. Normal, Conference by flash. 3. By refer, Conference by REFER method. 4. By star Number, Conference by star number (*1, *2, *3). 5. By flash number, Conference by flash number (0, 1, 2, 3, 4).
Conference server URL	Be used when the three way conference mode is “By refer”. In this mode the conference is hold by the specified conference server.
Dial in Conference function code	Default”*88”. To enable dial in conference function
Remove participant from Conference function code	To enable remove participant from Conference function code
Attended Call transfer	1. Disable (Default) 2. Enable
Call waiting temp de-active code	Default “*70”. To disable call waiting function for current call if Call Waiting is enable. After the user On Hook the function reconverts.
Call ID block temp active code	Default “*67”. To block call ID for current call ever the user “Call ID block” enable or disable. After the user On Hook the function reconverts.
Always forward active prefix number	Default “*72”. To inform the Proxy to do Always forward for current call and append the function code(“*72”) to front of the dial string. After the user On Hook the function reconverts.
Always forward de-active prefix number	Default “*73”. To inform the Proxy to cancel Always forward.
Call return function code	*69(Default) Defines the call return function code and NULL means disable.
Call redial function code	*68(Default) Defines the call redial function code and NULL means disable.
Unattended call transfer function code	Default “*66”. (See Table 3-30 - Unattended Call Transfer Operation)
Call Park function code	Default “*61”. User A Call User B, User A hold User B, User A dial “*61”

	to Park User B to Park server, After that User A hear busy tone to On Hook. Now the call User B with A has been Park.
Call Park server URL	To specify a Park server. Example “park@192.168.1.222”. If without the configuration auto generate a local park server in the device.
Call Park server share with tcid	1~8(2908) or 1~16(2916) or 1~24(2924) or 1~32(2932), The local park server share the TCP/IP port with a specified TCID.
Call pick up function code	Default “*62”. When there are Call be Park in the device a user (User C) can dial “*62” to pickup the call.
Call intercept function code	Default “*63”. When User A call to User B, User B is ringing. If User C in a same device with User B, User C dial “*63” to intercept the call. After that User B stop ring and conversion A and C.
Fax Signaling Method	Determines the SIP signaling method used to establish and convey a fax session after a fax is detected. 1. No fax negotiation using SIP signaling, changing code as configuration (G711 transparent mode). 2. Initiates T.38 fax relay.(Default) 3. Initiates fax using the coder G.711 A-law/μ-law. 4. Fax Fallback, Initiates T.38 fax relay. If the T.38 negotiation fails, the gateway re-initiates a fax session using the coder G.711 A-law/μ-law. Note 1: Fax adaptations: Echo Canceller = On Silence Compression = Off Echo Canceller Non-Linear Processor Mode = Off Note 2: If the gateway initiates a fax session using G.711, a ‘gpmd’ attribute is added to the SDP in the following format: For A-law: ‘a=gpmd:0 vbd=yes;ecan=on’. For μ-law: ‘a=gpmd:8 vbd=yes;ecan=on’.
T30 fax codec	1. G711 μ-law 2. G711 A-law
Fax tone detection	Fax tone detection and send reInvite to build fax communication. 1. Disable (Default) 2. Callee 3. Caller

	4. Both callee && caller
Modem Signaling Method	Determines the SIP signaling method used to establish and convey a data session after a modem tone is detected. 1. No negotiation using SIP signaling, changing code as configuration (G711 transparent mode). 2. Initiates data using the coder G.711 A-law/μ-law. Note 1: Fax adaptations: Echo Canceller = Off Silence Compression = Off Echo Canceller Non-Linear Processor Mode = Off Note 2: If the gateway initiates a fax session using G.711, a 'gpmd' attribute is added to the SDP in the following format: For A-law: 'a=gpmd:0 vbd=yes;ecan=off'. For μ-law: 'a=gpmd:8 vbd=yes;ecan=off'.
Modem codec	1. G711 μ-law 2. G711 A-law
Modem tone detection	Modem tone detection and send reInvite to build modem communication. 1. Disable (Default) 2. Callee 3. Caller 4. Both callee && caller
Report Flash	1. Disable (Default) 2. Enable, Report the flash hook event by re-invite request.
SDP Offans Only	1. Disable (Default) 2. Enable, Not to offer the only one codec with SDP again when receive more codec in the answer SDP.
Register Expire mode	1. Disable (Default), Tainet mode(Registration Expire = Expire * Re-registration Timing) 2. Enable, Huawei Mode(Registration Expire = Expire -600s(Expire >=1200s); Registration Expire = Expire/2(Expire <1200s)).
Request Re-Auth mode	1. Disable (Default), If receive 401/407 for a request with the Auth act it as a error code and not resend the request. 2. Enable, If receive 401/407 for a request with the Auth update the Auth parameters and send the request again.
Next Nonce mode	1. Disable (Default), not support the Next Nonce. 2. Enable, support the Next Nonce.

FXO auto route (2908 4S 4O only)	(Only for 4FXS 4FXO Type) 1. Disable (Default) 2. Enable, If a FXS Port dial to a not found number fail, auto route the call out from the first spare FXO.
Internal Dialing mode	1. Disable (Default) 2. Fix Number, enable internal dialing with the preset number (01, 02, 03….etc) 3. Tel Number, enable internal dialing with user specified Tel Number
Play Rapid Tone	1. Disable (Default) 2. Enable

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

Table 3-31 Call Waiting Operation

Call Waiting	Call Hold	Three way conference mode	Call Waiting Operation
Disable	*	*	(Not support)
Enable	Disable	*	(Not support)
Enable	Enable	Disable	Flash Hook
		Normal	Flash Hook
		By refer	Flash Hook
		By star number	Flash Hook, *2
		By flash number	Flash Hook, 2

Table 3-32 Call Conference normal Operation

Scene	Operation	Effect
Conversion A and B, A Hold B, A dial C, C Hook off	A Flash Hook	Three way conference
Conversion A and B, C dial A	A Flash Hook	A hold B; Conversion A and C;
Conversion A and B, C dial A, A Flash Hook,	A Flash Hook Again	Three way conference
Conversion A and B, A Hold B, A dial C, C Hook off, A Flash Hook, Do three way conference	A Flash Hook	A Bye C, Conversion A and B

Conversion A and B, C dial A, A Flash Hook, Do three way conference	A Flash Hook	A Bye C, Conversion A and B

Table 3-33 Call Conference By refer Operation

Scene	Operation	Effect
Conversion A and B, A Hold B, A dial C, C Hook off	A Flash Hook	A Call ConvServ(Conference server URL), After recv 200, Send Refer To B, To C; B Call ConvServ; C Call ConvServ; B and C Send Notify with terminal to A; A Bye B, A Bye C; A,B,C and ConvServ in a conference;
Conversion A and B, C dial A	A Flash Hook	A Call ConvServ, After recv 200, Send Refer To B, To C; B Call ConvServ; C Call ConvServ; B and C Send Notify with terminal to A; A Bye B, A Bye C; A,B,C and ConvServ in a conference;
A,B,C and ConvServ in a conference;	A Hook On	A Bye ConvServ(Conference server URL)
A,B,C and ConvServ in a conference;	B Hook On	B Bye ConvServ(Conference server URL)
A,B,C and ConvServ in a conference;	C Hook On	C Bye ConvServ(Conference server URL)
A,B,C and ConvServ in a conference;	ConvServ Bye A	A is kicked out the Confernence
A,B,C and ConvServ in a conference;	ConvServ Bye B	B is kicked out the Confernence
A,B,C and ConvServ in a conference;	ConvServ Bye C	C is kicked out the Confernence

Table 3-34 Call Conference By star number Operation

Scene	Operation	Effect
Conversion A and B, A Hold B, A dial C, C Hook off		A hold B; Conversion A and C;
A hold B; Conversion A and C;	A Flash Hook, dial “*1”	A hold C; Conversion A and B;
A hold C; Conversion A and B;	A Flash Hook, dial “*2”	A hold B; Conversion A and C;
Conversion A and B, A Hold B, A dial C, C Hook off	A Flash Hook, dial “*3”	Do three way conference.
Conversion A and B, C dial A	A Flash Hook, dial “*2”	A hold B; Conversion A and C;
A hold B; Conversion A and C;	A Flash Hook, dial “*1”	A hold C; Conversion A and B;
A hold C; Conversion A and B;	A Flash Hook, dial “*2”	A hold B; Conversion A and C;
Conversion A and B, C dial A	A Flash Hook, dial “*3”	Do three way conference.
A(organizer), B, C in a conference.	A Hook On	A Bye B; A Bye C;
A(organizer), B, C in a conference.	B Hook On	B Bye A; Conversion A and C;
A(organizer), B, C in a conference.	C Hook On	C Bye A; Conversion A and B;

Call Conference By flash number

There are five function codes to be used in Venus 29xx.

“0” – Disconnect the last call.

“1” – Disconnect the previous call.

“2” – Switch call or be used to accept the waiting call.

“3” – Do three way conferences.

“4” – Call transfer by flashhook + function code “4” when “Attended Call transfer” is set to enable.

Table 3-35 Call Conference By flash number Operation

	Call Switch	Call Hold	Call Waiting	Conferences by Call Hold	Conferences by Call Waiting
	A hold B A and C converse	A and B converse A FLASHES A dials C	A and B converse C dials A	A and B converse A hold B,A call C A and C converse A,B,C converse	A and B converse C call A A and C converse A,B,C converse
“ 0”	Bye C	-	Terminate C	Bye C	Bye C
“ 1”	Bye B	-	Bye B	Bye B	Bye B
“ 2”	A and B converse C gets MOH A and C converse B gets MOH	-	A and C converse B gets MOH	A and C converse B gets MOH	A and B converse C gets MOH
“ 3”	A,B,C converse	-	A,B,C converse	-	-
“ 4”	B and C converse C send BYE to A A receives busy tone	C send 487 to A A receives busy tone C ring	-	-	-

Table 3-36 Attended Call Transfer Operation

Attended Call transfer	Three Way conference mode	Operation
Disable	*	(Not support)
Enable	By flash number mode	Organizer Flash Hook, Dial “4”
Enable	Other mode	Organizer Hook On

Table 3-37 Unattended Call Transfer Operation

Scene	Operation	Effect
Conversion A and B, A Hold B	A dial Unattended Call Transfer function code(*66); A play Stutter Tone; A dial C number	A play Busy Tone(and A Hook On), C Ring, C Hook Off, Conversion C and B;

3.5.5.3 Region Profile Configuration

The Regional configuration is used to assign specified regional profile, which contains Ring and Tone definition. Also the country code, area code, internal access code and domestic access code are also defined here. These parameters are used when E.164 function is enabled.

1) Region

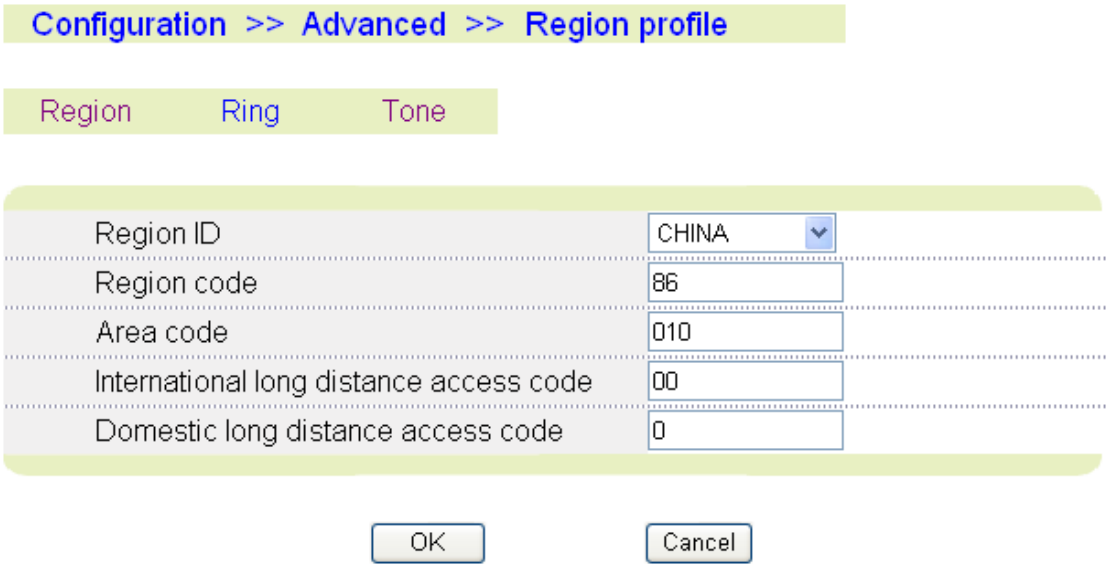


Figure 3-48 Region Configuration

Table 3-38 Description of Region Configuration Items

Field	Description
Region ID	The region ID of profile. 0 = GENERIC 1 = USA 2 = CHINA 3 = TAIWAN 4 = HONGKONG 5 = INDIA 6 = SWEDEN 7 = MALAYSIA 8 = FRENCH 9 = CZECH 10 = SLOVAKIA 11 = AUSTRALIA 12 = POLAND 13 = BULGARIA 14 = NEWZEALAND
Region code	The Country/Region code
Area code	The area code
International long distance access code	The international access code
Domestic long distance access code	The domestic access code

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

2) Ring

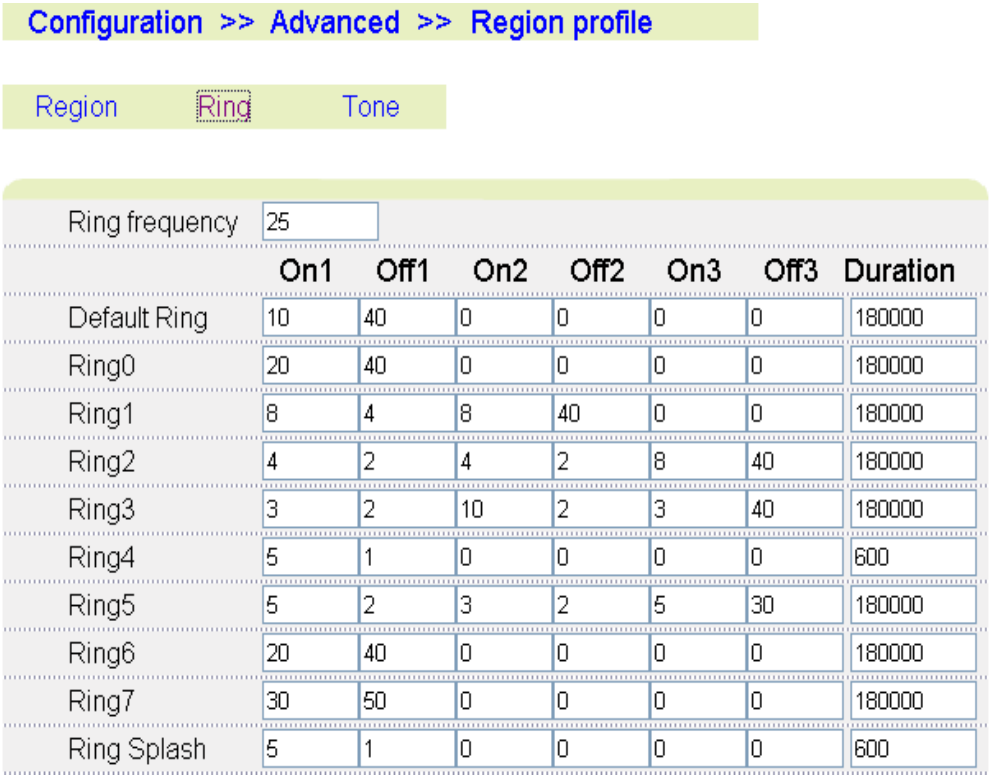


Figure 3-49 Ring Configuration

Table 3-39 Description of Ring Configuration Items

Field	Description
Ring frequency	25 (Default) Ring frequency and different with different region ID, valid value is 17 to 50.
Default Ring Ring 0 ~ Ring 7 Ring Splash	Ring cadence On 1 in 100ms. Note: index 1-8 means distinctive ring R0-R7, index 9 means ring splash.

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3) Tone

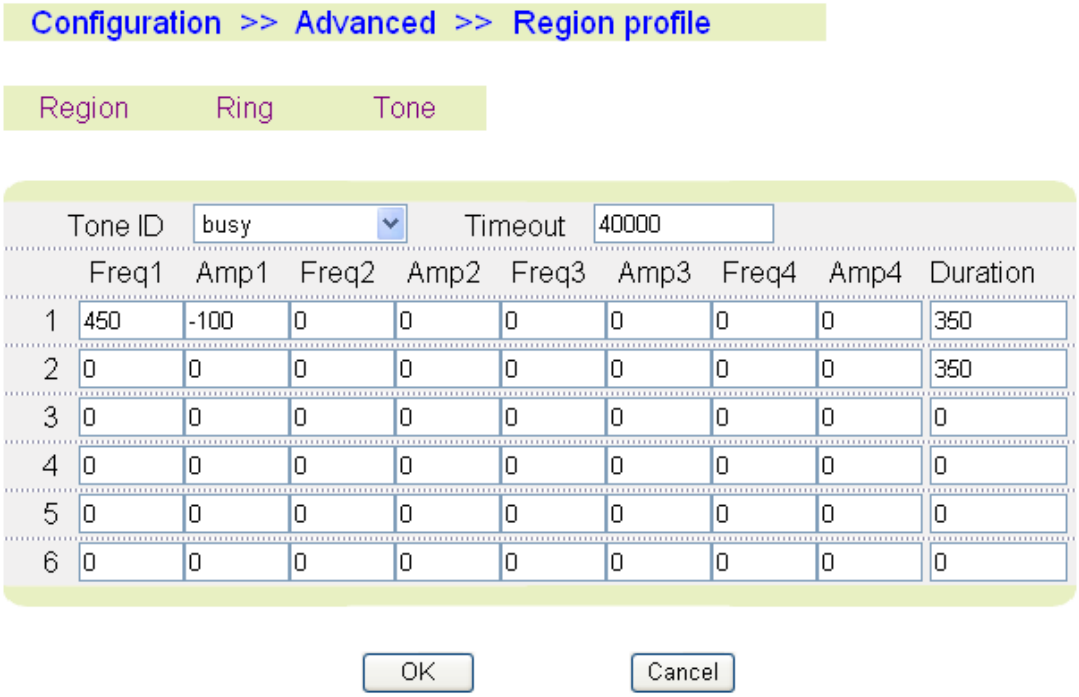

Configuration >> Advanced >> Region profile
Region
Ring
Tone
Tone ID
busy
Timeout
40000
Freq1 Amp1 Freq2 Amp2 Freq3 Amp3 Freq4 Amp4 Duration
1 450 -100 0 0 0 0 0 0 350
2 0 0 0 0 0 0 0 0 350
3 0 0 0 0 0 0 0 0 0
4 0 0 0 0 0 0 0 0 0
5 0 0 0 0 0 0 0 0 0
6 0 0 0 0 0 0 0 0 0
OK
Cancel

Figure 3-50 Tone Configuration

Table 3-40 Description of Tone Configuration Items

Field	Description
Tone ID [0-17] [1-6]	Tone frequency 1. First index means tone id and second index means tone element. 0 = busy 1 = ring_back 2 = congestion 3 = dial_tone 4 = dial_tone2 5 = out_of_service 6 = addr_ack 7 = disconnect 8 = offhook_notice 9 = offhook_alert 10 = dial_tone3 11 = off_on_dial_tone 12 = call_wait 13 = reorder 14 = keypad_echo1 15 = rapid_tone

	16 = rmt_disconnect 17 = call_hold
Timeout [0-17]	Default tone timeout in ms.
Freq1 [0-17][1-6]	Tone frequency 1.
Amp 1 [0-17][1-6]	Tone amplitude 1, in 1/10db.
Freq 2 [0-17][1-6]	Tone frequency 2.
Amp 2 [0-17][1-6]	Tone amplitude 2, in 1/10db.
Freq 3 [0-17][1-6]	Tone frequency 2.
Amp 3 [0-17][1-6]	Tone amplitude 3, in 1/10db.
Freq 4 [0-17][1-6]	Tone frequency 2.
Amp 4 [0-17][1-6]	Tone amplitude 4, in 1/10db.
Duration [0-17][1-6]	Tone element duration.

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.5.4 DDNS

Dynamic DNS(DDNS) is a method, protocol or network service that provides the capability for a network device using the internet Protocol Suite, such as an IP router or computer system, to notify a domain name server to change, in real time(ad-hoc), the active DNS configuration of its configured hostnames, addresses or other information stored in DNS. Through this technique, Our VENUS 29xx can communicate with each other just by setting the peer domain name.

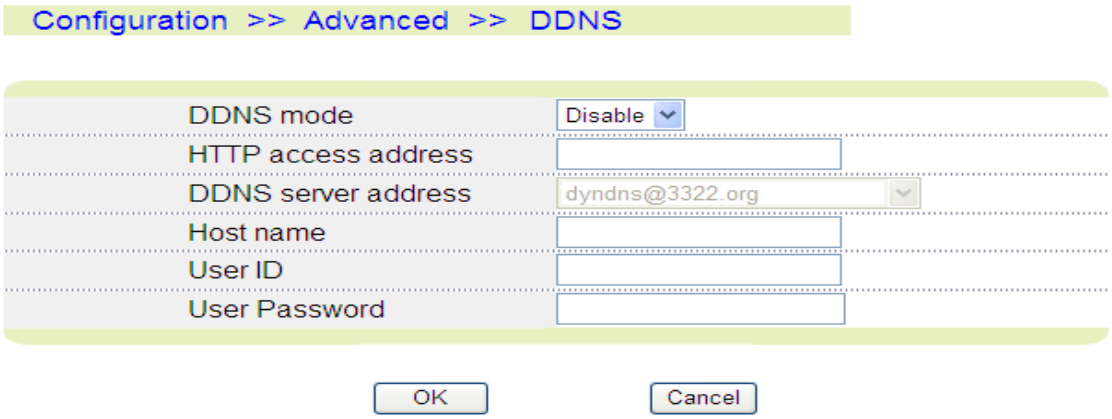


Figure 3-51 DDNS Configuration

Table 3-41 Description of DDNS Configuration Items

Field	Description
DDNS mode	Default: Disable Enable, start up DDNS function.
HTTP access address	The HTTP server IP address , this is used for private network pass through when VENUS 29xx is in private network. This may be provided by DDNS service provider.
DDNS server address	Dynamic DNS server address , This is used for update locate active information such as Local IP address
Host name	The domain name in DDNS
User ID	The user ID use for authentication
User password	The password use for authentication

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.5.5 NAT Traversal Configuration

User may assign one NAT address if the NAT server is DMZ. Otherwise, STUN should be used. Simple Traversal of UDP through NATs (STUN) (according to RFC 3489) is a client/server protocol that solves most of the NAT traversal problems. The STUN server operates in the public Internet and the STUN clients are embedded in end-devices (located behind NAT). STUN is used both for the signaling and the media streams. STUN works with many existing NAT types, and does not require any special behavior from them.

STUN enables the gateway to discover the presence (and types) of NATs and firewalls located between it and the public Internet. It provides the gateway with the

capability to determine the public IP address and port allocated to it by the NAT. This information is later embedded in outgoing SIP/SDP messages and enables remote SIP user agents to reach the gateway.

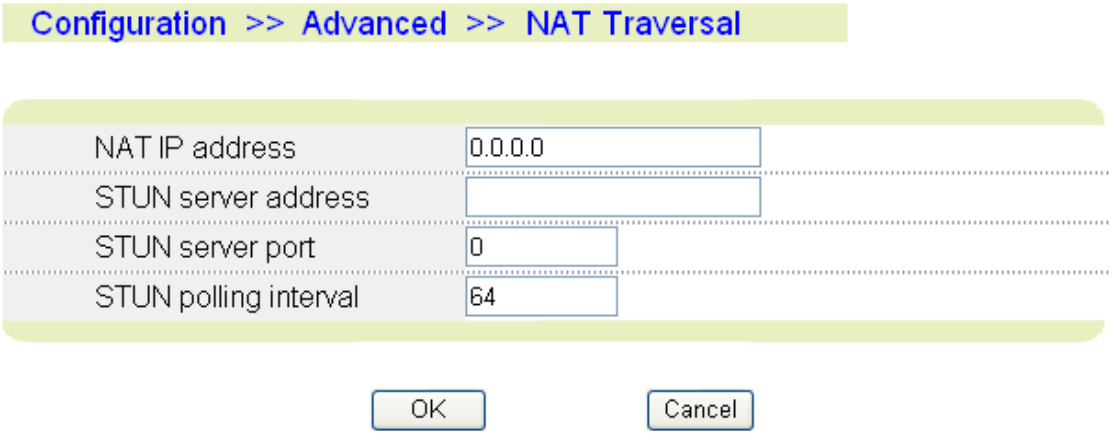


Figure 3-52 NAT Traversal Configuration

Table 3-42 Description of NAT Traversal Configuration Items

Field	Description
NAT IP address	The external IP address of DMZ NAT server. Disable this function to assign IP address of 0.0.0.0.
STUN server address	Stun server IP address or domain name.
STUN server port	0(Default) Stun server service port number.
STUN polling interval	64 (Default) Defines the time interval (in seconds) the STUN client requests for a NAT keep alive update.

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.5.6 SLIC Configuration

SLIC Configuration is used to adjust the Current and Voltage Value of SLIC. By changing the current value, user may increase the distance between FXS port and telephone.

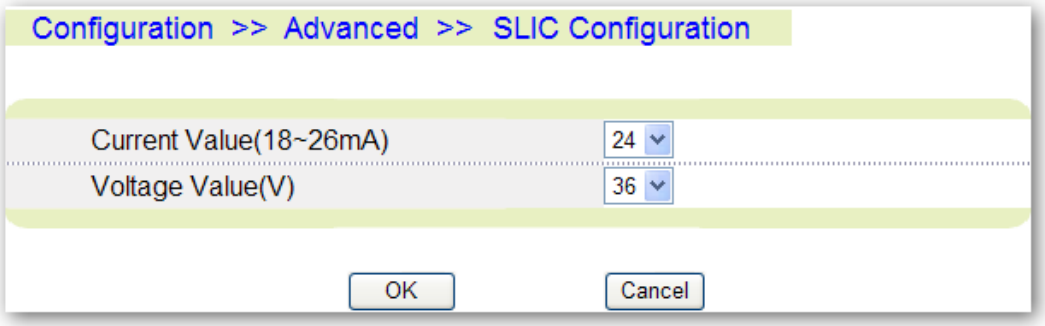


Figure 3-53 SLIC Configuration

Table 3-43 Description of SLIC Configuration Items

Field	Description
Current Value	Adjust the Current Value
Voltage Value	Select the appropriate Voltage Value

If the configuration is finished, click “OK” button below the web page to confirm changes or click “Cancel” button to deny it.

3.5.5.7 Meter Configuration

Meter Configuration is used to adjust the settings of metering tone generation.

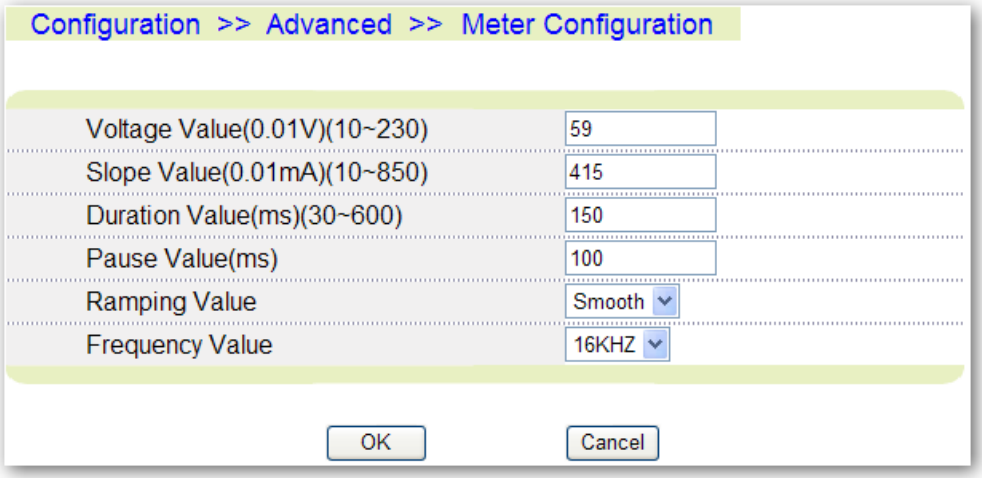


Figure 3-54 Meter Configuration

Table 3-44 Description of Meter Configuration Items

Field	Description
Voltage Value	Select the appropriate Voltage Value
Slope Value	Select the appropriate Slope Value

Duration Value	Duration time of metering tone
Pause Value	Pause time of metering tone
Ramping Value	1.Smooth 2.Abrupt
Frequency Value	1.16KHZ 2.12KHZ

3.5.6 Data Configuration

Table 3-45 The Data Configuration Menu Items

Menu	Description
Static Routing (Eth1 under Router mode)	Configuration static routing information which use for LAN
NAT mapping (Eth1 under Router mode)	Configuration NAT information which use for LAN
DHCP Server (Eth1 under Router mode)	Configuration DHCP server information which use for LAN
QOS	Configuration of IP QoS.
VLAN (Eth1 under Switch mode)	Configuration of VLAN.

3.5.6.1 Static Routing

Static Routing information is used for routing data between LAN and internet when LAN is in router mode.

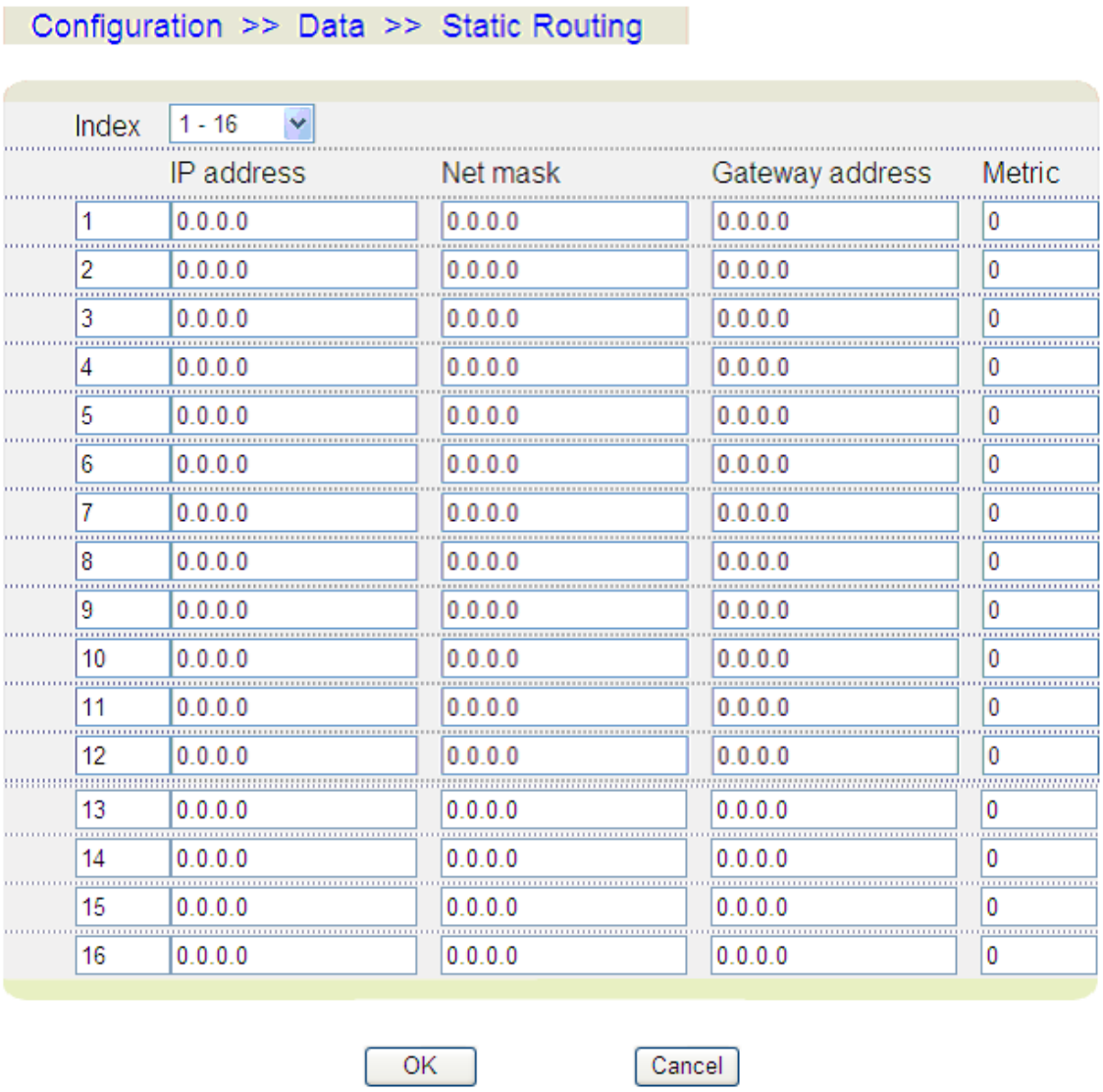


Figure 3-55 Static Routing Configuration

Table 3-46 Description of Static Routing Items

Field	Description
IP address	The network address of this route record
Net mask	The net mask of the network address
Gateway address	The gateway IP address of this route record
Metric	The metric use for routing data to next hop

3.5.6.2 NAT mapping

In computer networking, network address translation(NAT) is the process of modifying network address information in datagram packet headers while in transit across a traffic routing device for the purpose of remapping a given address space into another. The NAT has two class, SNAT(Source NAT) and DNAT(Destination NAT). Our VENUS 29xx just supports DNAT to transit packages between LAN and internet when LAN is in router mode.

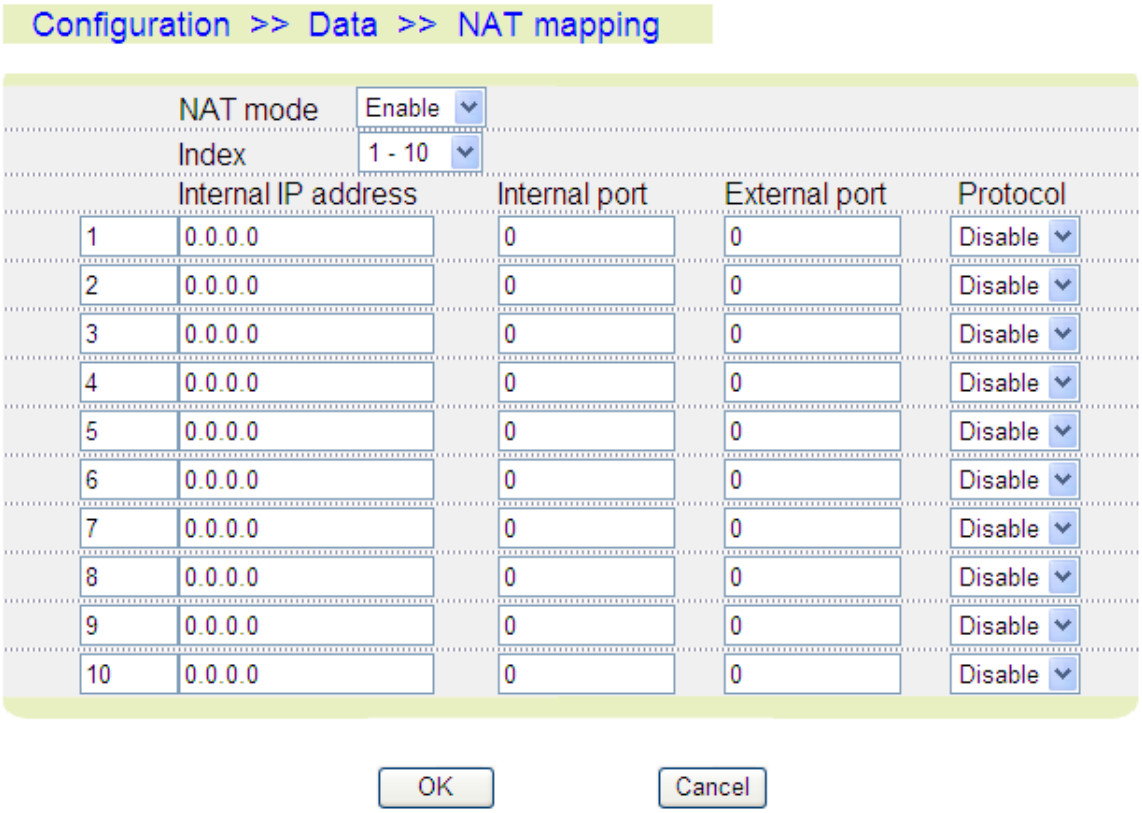


Figure 3-56 NAT mapping Configuration

Table 3-47 Description of NAT mapping Items

Field	Description
NAT mode	Default:disable Enable, start up NAT mode
index	The current NAT mapping records (1-50).
Internal IP address	The IP address of device in LAN
Internal port	The IP port of device in LAN
External port	The IP port of wan which is used for access through internet
Protocol	The NAT mapping protocol, Default disable TCP :mapping TCP protocol only UDP:mapping UDP protocol only

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.6.3 DHCP Server

DHCP service runs on a central server that dynamically assigns an IP address to individual PCs or workstations. This protocol delivers IP information on a LAN. The service can be used just when LAN is in router mode.

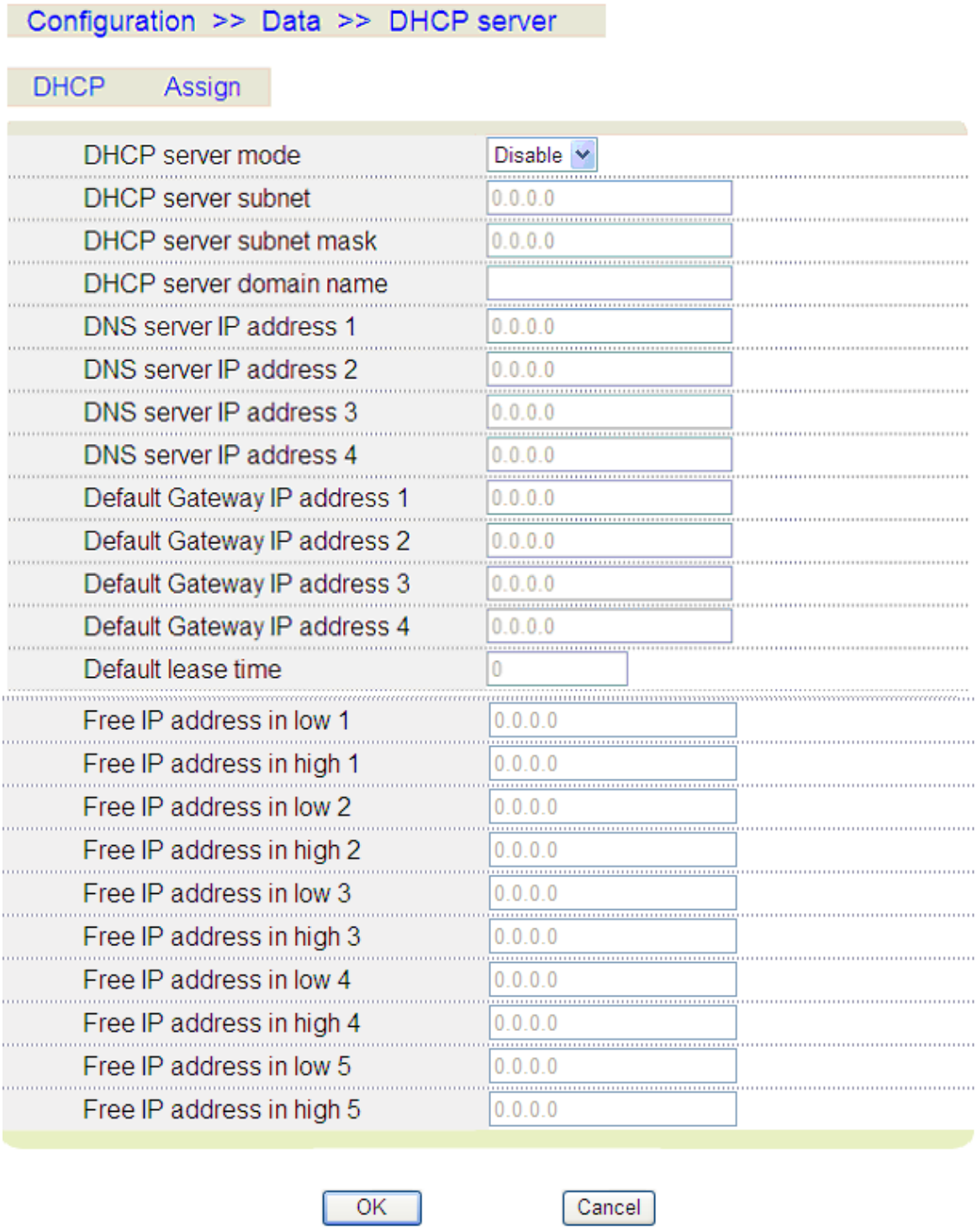


Figure 3-57 DHCP Server Configuration

Table 3-48 Description of DHCP Server DHCP Items

Field	Description
DHCP server mode	Default: disable Enable, start up DHCP service
DHCP server subnet	The subnet on which the DHCP service run
DHCP server subnet mask	The subnet mask of the subnet
DHCP server domain name	The DHCP server domain name
DNS server IP address 1-4	The DNS server IP address use for LAN, you can configure four DNS server
Default gateway IP address 1-4	The default gateway IP address for LAN, you can configure four gateway
Default lease time	The lease time for IP assign for device in LAN
Free IP address in low 1-5	The low and high consist of a range for IP assign. You can configure four IP range.
Free IP address in high 1-5	The low and high consist of a range for IP assign. You can configure four IP range.

Figure 3-58 DHCP Server Configuration

Table 3-49 Description of DHCP Server Assign Items

Field	Description
IP address	This IP is assigned to a static IP
Host name	The host name of the static IP
MAC address	The MAC address of the static IP

3.5.6.4 QoS Configuration

DiffServ is architecture providing different types or levels of service for IP traffic. DiffServ (according to RFC 2474) offers the capability to prioritize certain traffic types, depending on their priority, thereby accomplishing a higher-level QoS at the

expense of other traffic types. By prioritizing packets, DiffServ routers can minimize transmission delays for time-sensitive packets such as VoIP packets.

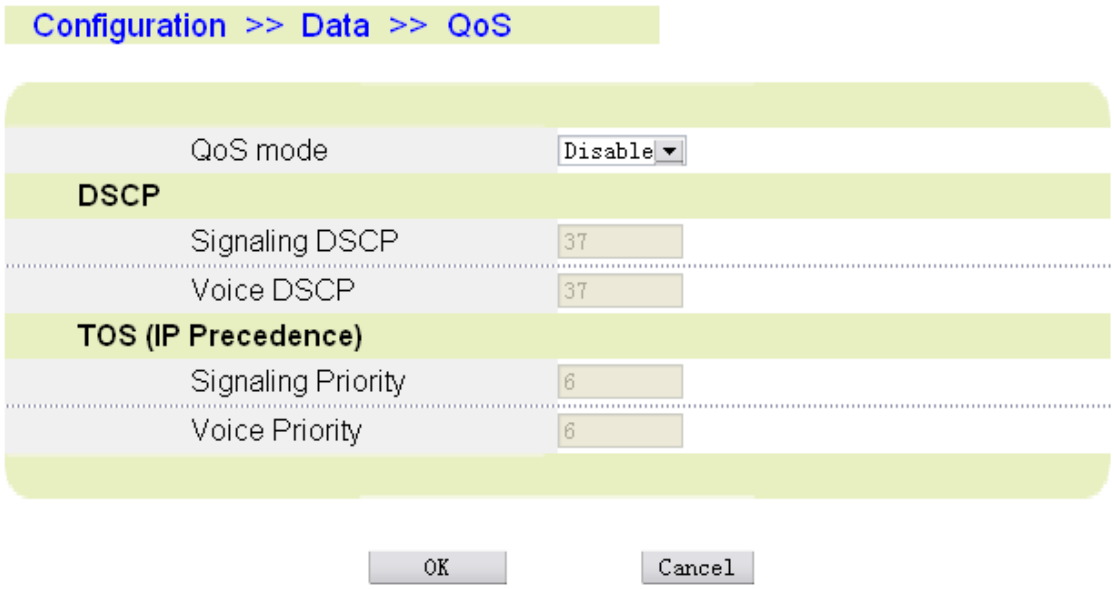


Figure 3-59 QoS Configuration

Table 3-50 Description of QoS Items

Field	Description
QoS mode	1.Disable 2.DSCP 3.TOS (IP precedence)
Signaling DSCP	The signaling DSCP. Default is 37.
Voice DSCP	The voice DSCP. Default is 37.
Signaling TOS	The signaling TOS. Default is 6.
Voice TOS	The voice TOS. Default is 6.

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.5.6.5 VLAN Configuration

Venus 29xx only support port base VLAN. It support tag for port also.

There are three groups for VLAN setting. You can set any group for your need. Zero to six ports can be in one VLAN. If the setting is repeated it will be ignored.

The “Port Based” page is used to partition VLAN and the “Tag Based” page to set VLAN tag.

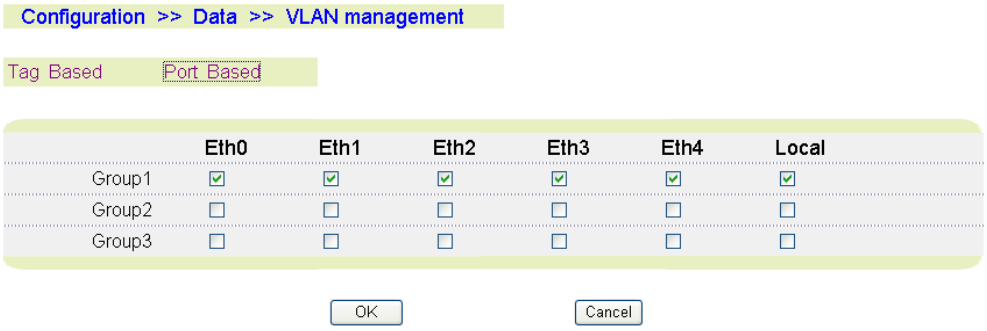


Figure 3-60 Port Based Vlan Configuration

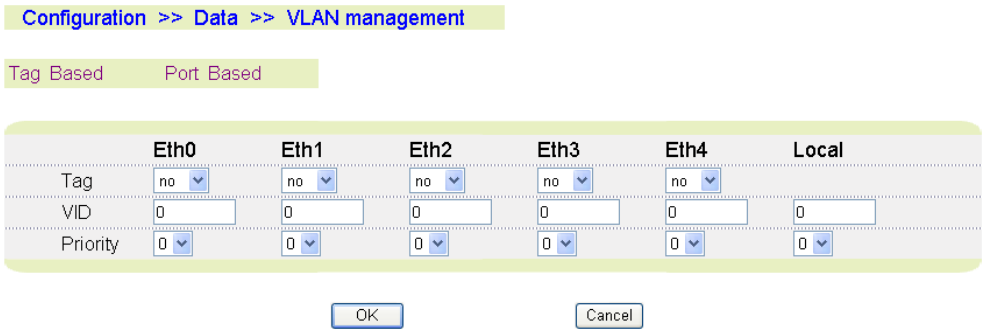


Figure 3-61 Vlan Tag Configuration

Table 3-51 Description of Vlan Items

Field	Description
VLAN Group[6]	Eth0, Eth1, Eth2~4, local. Default no select.
VLAN Tag[5]	Eth0, Eth1or Eth2~4 1. no 2.yes
VLAN Priority[6]	Default 0. (Notes: Current no using)
VLAN VID[6]	Eth0, eth1, Eth2~4, local 0~4094

If the configuration is finished, click “OK” button bellow the web page to confirm changes or click “Cancel” button to deny it.

3.6 Maintenance

The Maintenance menu is used to manage the system state and the firmware. It is divided into four groups, “Fault management”, “Performance Management”, “Diagnostics”, and “Device Control”.

Table 3-52 The Maintenance Menu Items

Menu	Description
Fault Management	FM Monitoring and Clear FM Records.
Performance Management	Ethernet, Call Control, RTP, SIP Agent, SIP Call Initiation, NAT, DHCP Server and Clear PM Records.
Diagnostics	Do Ping, STUN test and Traceroute.
Device Control	F/M Upgrade, Config Up/Down, Syslog Upload, Reset to Default and Reboot.

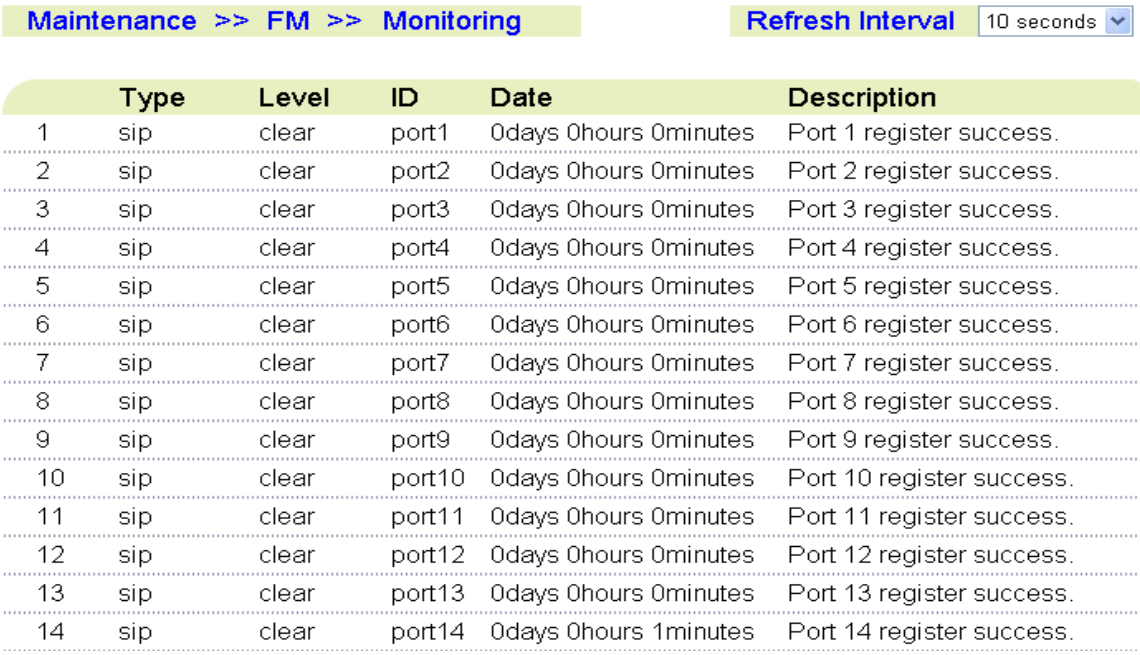
3.6.1 Fault Management

Table 3-53 The Fault Management Menu Items

Menu	Description
Monitoring	Fault alarm messages monitoring.
Clear Records	Fault alarm messages cleaning.

3.6.1.1 Monitoring

There are six levels for alarm message, “Clear”, “Event”, “Warning”, “Minor”, “Major” and “Critical”.



	Type	Level	ID	Date	Description
1	sip	clear	port1	0days 0hours 0minutes	Port 1 register success.
2	sip	clear	port2	0days 0hours 0minutes	Port 2 register success.
3	sip	clear	port3	0days 0hours 0minutes	Port 3 register success.
4	sip	clear	port4	0days 0hours 0minutes	Port 4 register success.
5	sip	clear	port5	0days 0hours 0minutes	Port 5 register success.
6	sip	clear	port6	0days 0hours 0minutes	Port 6 register success.
7	sip	clear	port7	0days 0hours 0minutes	Port 7 register success.
8	sip	clear	port8	0days 0hours 0minutes	Port 8 register success.
9	sip	clear	port9	0days 0hours 0minutes	Port 9 register success.
10	sip	clear	port10	0days 0hours 0minutes	Port 10 register success.
11	sip	clear	port11	0days 0hours 0minutes	Port 11 register success.
12	sip	clear	port12	0days 0hours 0minutes	Port 12 register success.
13	sip	clear	port13	0days 0hours 0minutes	Port 13 register success.
14	sip	clear	port14	0days 0hours 1minutes	Port 14 register success.

Figure 3-62 FM Monitoring

Table 3-54 Description of FM Monitoring

Field	Description
Refresh Interval	1. disable 2. 10 s(Default) 3. 20 s 4. 30 s 5. 40 s 6. 50 s 7. 60 s Refresh the alarm messages.

3.6.1.2 Clear Records

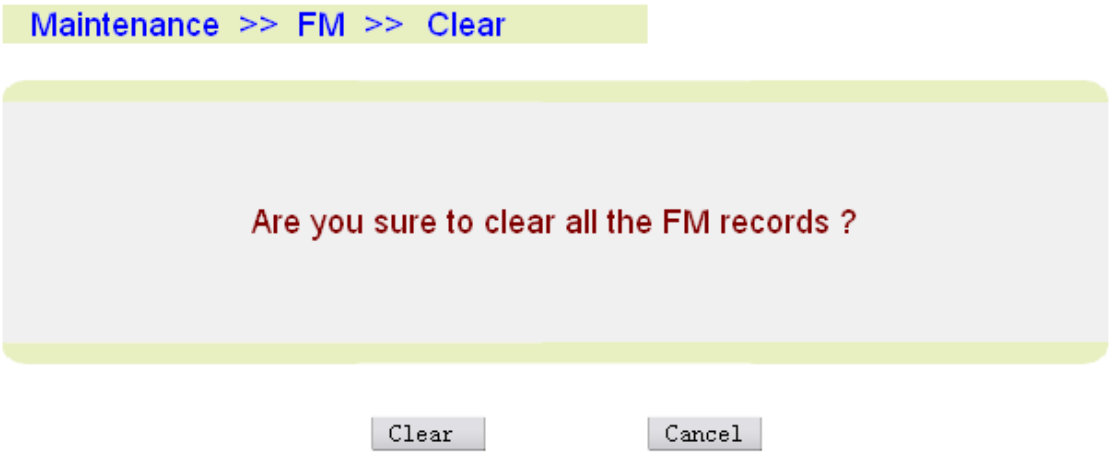


Figure 3-63 FM Clear

3.6.2 Performance Management

Table 3-55 The Performance Management Menu Items

Menu	Description
Ethernet	Ethernet status
Call Control	Call Control status
RTP	RTP status
SIP Agent	SIP Agent status
SIP Call Initiation	SIP Call Initiation status
Clear	Performance messages cleaning.

3.6.2.1 Ethernet

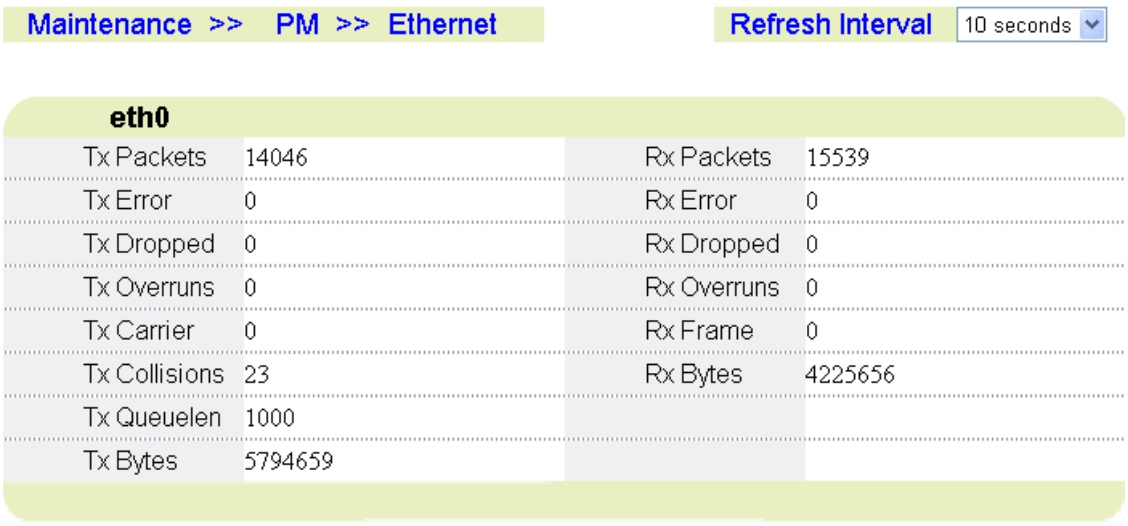
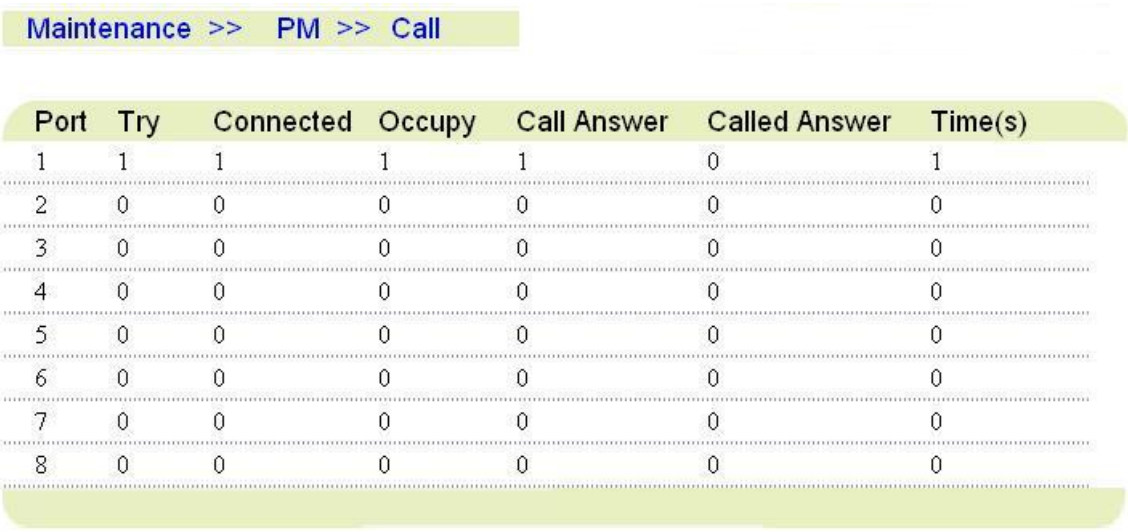


Figure 3-64 Ethernet Performance

3.6.2.2 Call Control



Port	Try	Connected	Occupy	Call Answer	Called Answer	Time(s)
1	1	1	1	1	0	1
2	0	0	0	0	0	0
3	0	0	0	0	0	0
4	0	0	0	0	0	0
5	0	0	0	0	0	0
6	0	0	0	0	0	0
7	0	0	0	0	0	0
8	0	0	0	0	0	0

Figure 3-65 Call Control Status

3.6.2.3 RTP

Maintenance >> PM >> RTP Refresh Interval 10 seconds

Port	Drop(pkts)	toNet(pkts)	toNet(bytes)	toDSP(pkts)	toDSP(bytes)
1	0	0	0	0	0
Last1	0	0	0	0	0
2	0	0	0	0	0
Last2	0	0	0	0	0
3	0	0	0	0	0
Last3	0	0	0	0	0
4	0	0	0	0	0
Last4	0	0	0	0	0
5	0	0	0	0	0
Last5	0	0	0	0	0
6	0	0	0	0	0
Last6	0	0	0	0	0
7	0	0	0	0	0

Figure 3-66 RTP Status

3.6.2.4 SIP Agent

Maintenance >> PM >> SIP agent

Transactions	75
Rx invite messages	0
Rx invite retransmission messages	0
Rx non-invite messages	0
Rx non-invite retransmission messages	0
Rx respones	158
Rx response retransmission	8
Tx invite messages	2
Tx invite retransmission messages	0
Tx non-invite messages	79
Tx non-invite retransmission messages	4
Tx respones	0
Tx response retransmission	0

Figure 3-67 SIP Agent Status

3.6.2.5 SIP Call Initiation

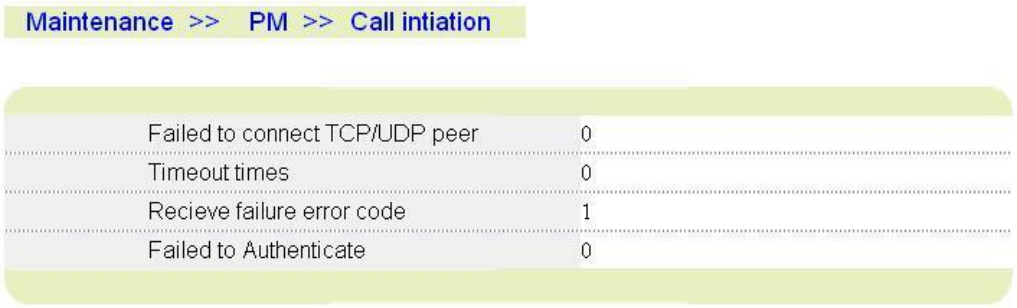

Maintenance >> PM >> Call intiation
Failed to connect TCP/UDP peer 0
Timeout times 0
Recieve failure error code 1
Failed to Authenticate 0

Figure 3-68 SIP Call Initiation Status

3.6.2.6 Clear

Figure 3-69 Clear Performance Message

3.6.3 Diagnostics

Table 3-56 The Fault Management Menu Items

Menu	Description
Ping	Ping the specify IP device.
Stun	Check a Simple Traversal of User Datagram Protocol Through Network Address.
Traceroute	To determine the route taken by packets across the IP network

3.6.3.1 Ping

This feature can be used to verify the path of packet transmission between VENUS 29xx Series and another node. For example, to check whether the Proxy Server is reachable or is alive. Make sure you configure the correct subnet mask, and the IP address of the default gateway.

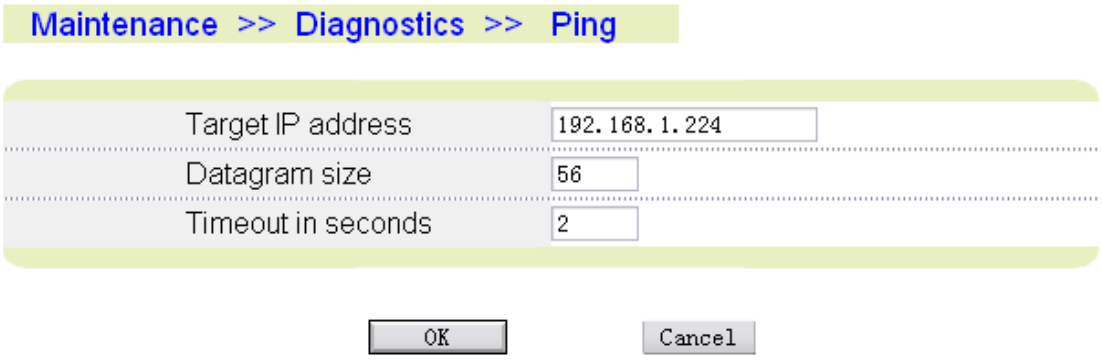


Figure 3-70 Ping

Click “OK” button below the web page to do or click “Cancel” button to deny it.

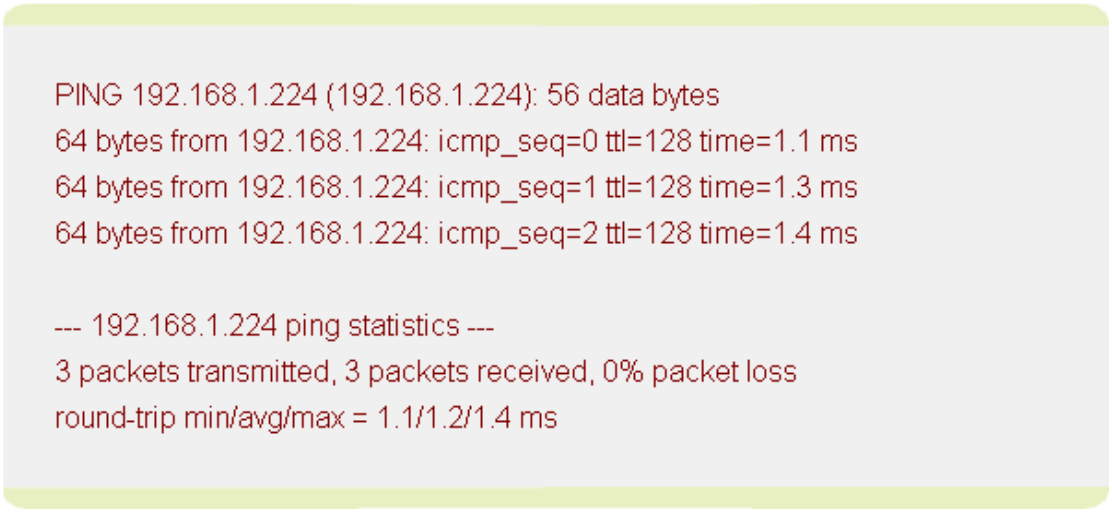

PING 192.168.1.224 (192.168.1.224): 56 data bytes
64 bytes from 192.168.1.224: icmp_seq=0 ttl=128 time=1.1 ms
64 bytes from 192.168.1.224: icmp_seq=1 ttl=128 time=1.3 ms
64 bytes from 192.168.1.224: icmp_seq=2 ttl=128 time=1.4 ms
--- 192.168.1.224 ping statistics ---
3 packets transmitted, 3 packets received, 0% packet loss
round-trip min/avg/max = 1.1/1.2/1.4 ms

Figure 3-71 result of ping a connected IP device

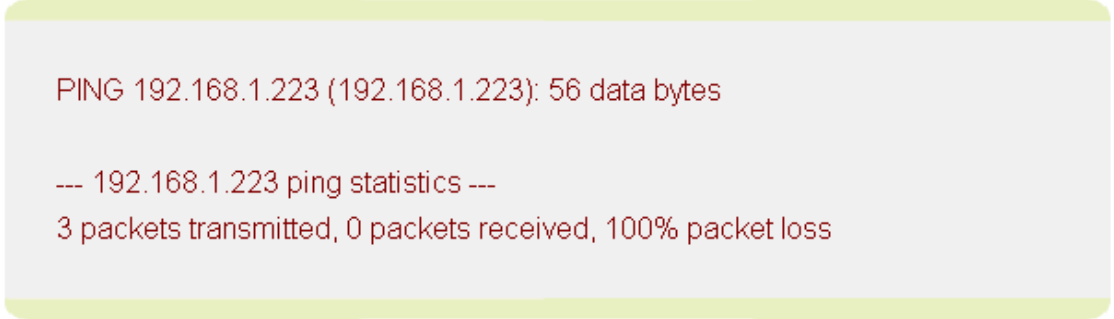

PING 192.168.1.223 (192.168.1.223): 56 data bytes
--- 192.168.1.223 ping statistics ---
3 packets transmitted, 0 packets received, 100% packet loss

Figure 3-72 result of ping a disconnected IP device

3.6.3.2 Stun

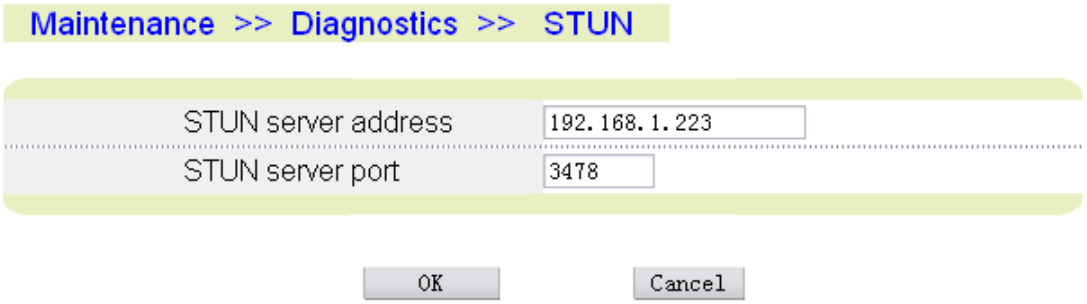


Figure 3-73 Stun

Click “OK” button bellow the web page to do or click “Cancel” button to deny it.

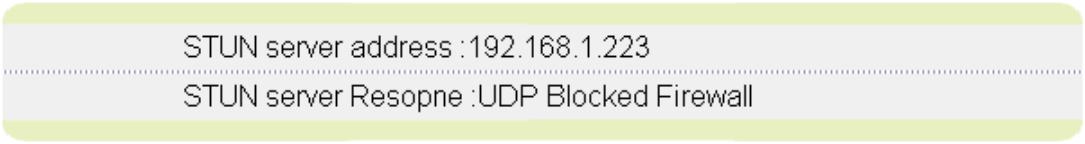


Figure 3-74 result of stun a disconnected Stun server

3.6.3.3 Traceroute

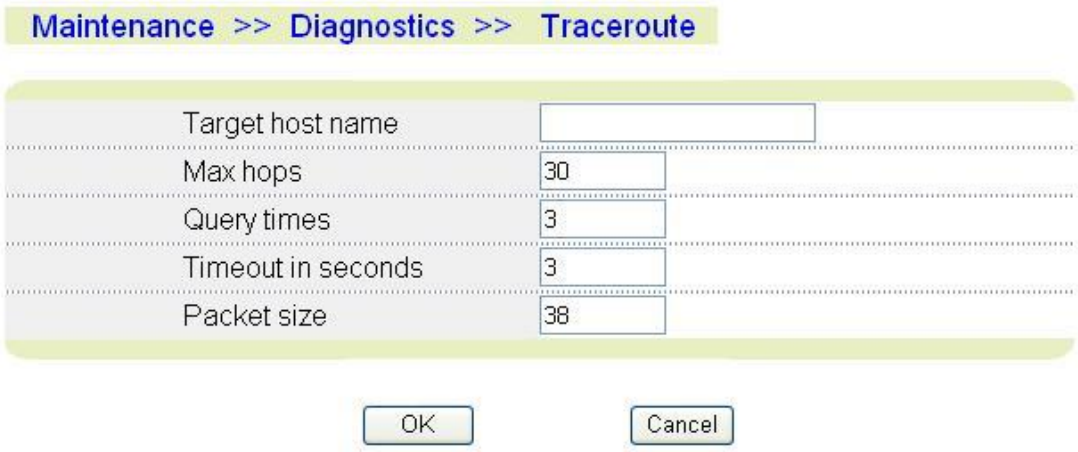


Figure 3-75 Traceroute

Click “OK” button below the web page to do or click “Cancel” button to deny it.

3.6.4 Device Control

Table 3-57 The Device Control Menu Items

Menu	Description
F/W Upgrade	Firm ware upgrade
Config Up/Down	Config up load and down load
Syslog Upload	Upload the recorded system log
Reset to Default	Restore the default configuration to the device.
Reboot	Reboot the device

3.6.4.1 F/W Upgrade

VENUS 2908 Series allows software upgrade for feature improvement. The new firmware (the Application Program), in binary code format, can be remotely downloaded from a TFTP server and written into the internal flash memory of the VENUS 2908 Series. Check with our technical support for the availability of the latest software.

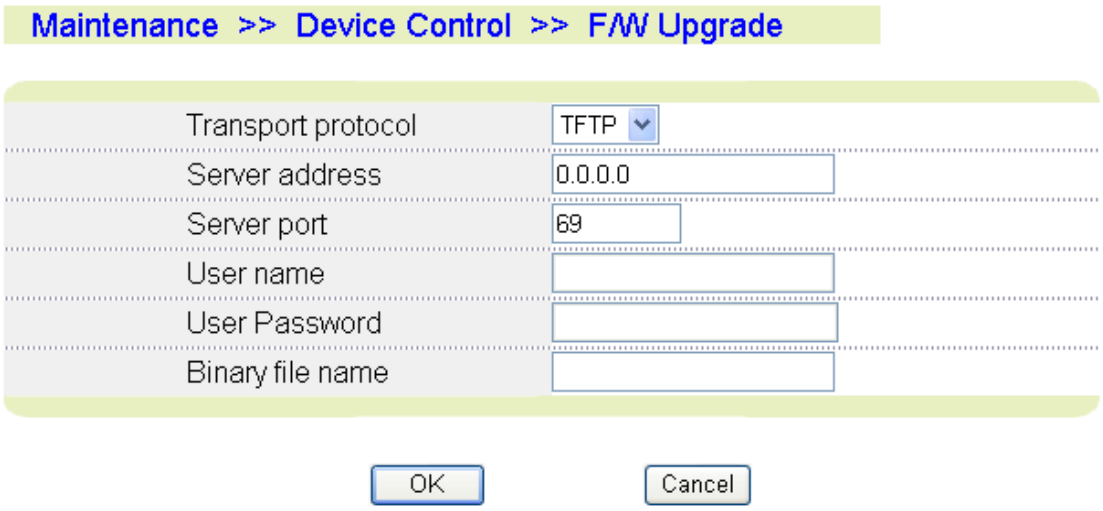


Figure 3-76 Firmware Upgrade

Table 3-58 Description of Firmware Upgrade Items

Field	Description
Transport protocol	Select the transport protocol
Server address	This item tells VENUS 29xx Series which IP address of TFTP server that includes the firmware.
Server port	This item tells VENUS 29xx Series which port of TFTP server that includes the firmware.
User name	User name of the server

Field	Description
Transport protocol	Select the transport protocol
User Password	User password of the server
Binary file name	This item tells VENUS 29xx Series that file should be loaded from TFTP server. The firmware must be in the binary format.

Step 1: Input server IP address and firmware file name.

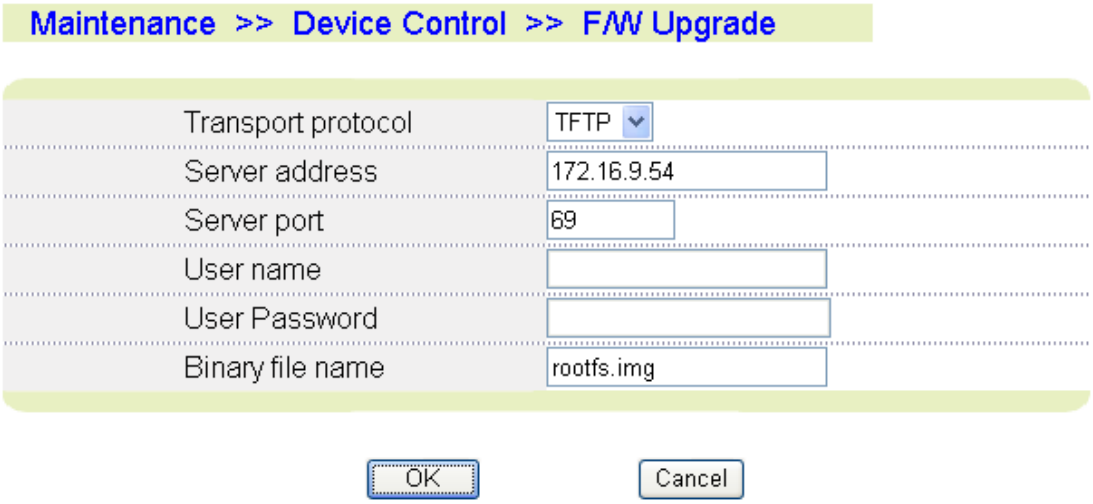


Figure 3-77 Step 1 of F/W Upgrading

Note:
You need to run a TFTP server (such as Cisco TFTP server) for the upgrading except the TFTP server of the server has run.

If the configuration is finished, click “OK” button below the web page to execute upgrade or click “Cancel” button to give up. The whole upgrading process will take about 140 seconds.

Caution:
Upgrade the software when there is no phone call in progress. Otherwise, VoIP service will be abruptly terminated

Step 2: Download

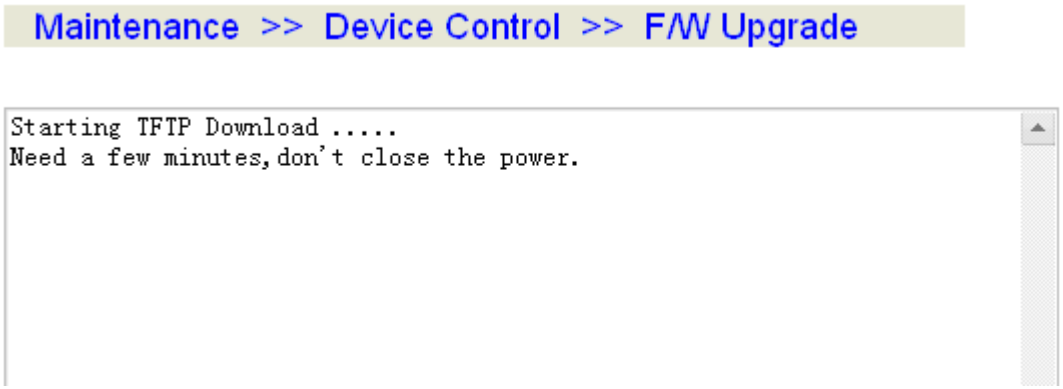


Figure 3-78 Step 2 of F/W Upgrading

The download process will take about 20 seconds.

If the TFTP server is not running or firmware file does not exist the upgrading will abort.

Caution:

Do not write the software with incorrect checksum and do not reset the device while performing flash write. In case the flash write is abnormally disrupted, VENUS Series can still boot up using the Boot code. Re-download the software and re-program the flash again to recovery the AP software

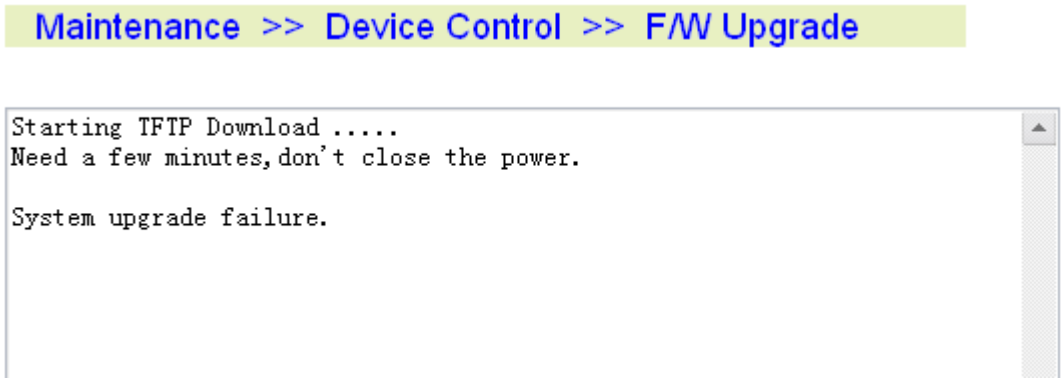

Maintenance >> Device Control >> F/W Upgrade
Starting TFTP Download .....
Need a few minutes,don't close the power.
System upgrade failure.

Figure 3-79 F/W Upgrading failure

Step 3: Write to Flash

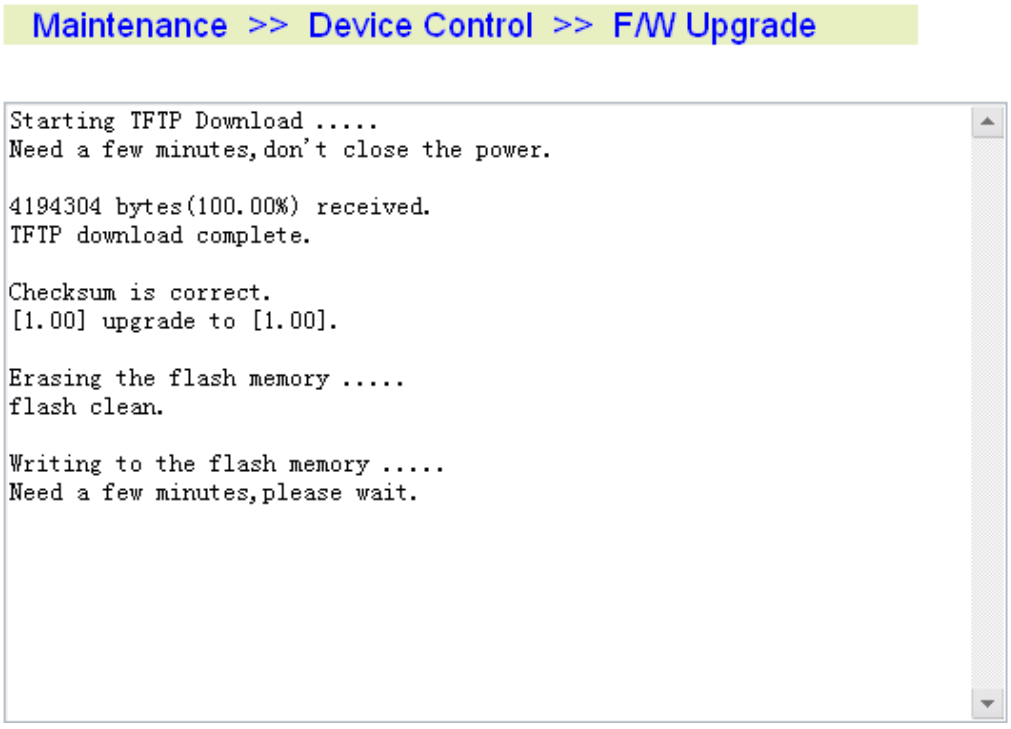

Maintenance >> Device Control >> F/W Upgrade
Starting TFTP Download .....
Need a few minutes,don't close the power.
4194304 bytes(100.00%) received.
TFTP download complete.
Checksum is correct.
[1.00] upgrade to [1.00].
Erasing the flash memory .....
flash clean.
Writing to the flash memory .....
Need a few minutes,please wait.

Figure 3-80 Step 3 of F/W Upgrading

The flash writing process will finish in 80 seconds after download.

Step 4: Reboot system

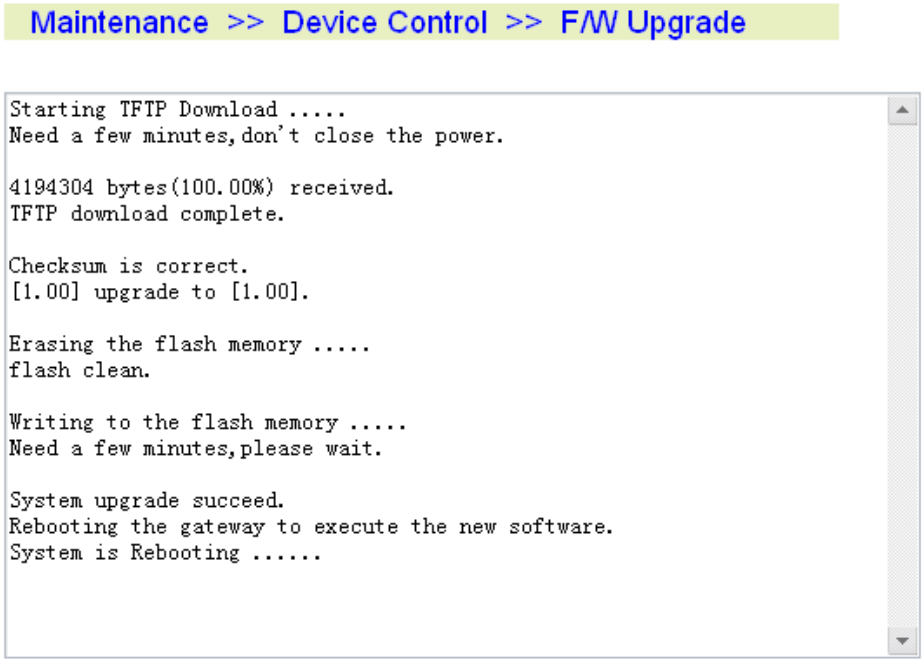

Maintenance >> Device Control >> F/W Upgrade
Starting TFTP Download .....
Need a few minutes,don't close the power.
4194304 bytes(100.00%) received.
TFTP download complete.
Checksum is correct.
[1.00] upgrade to [1.00].
Erasing the flash memory .....
flash clean.
Writing to the flash memory .....
Need a few minutes,please wait.
System upgrade succeed.
Rebooting the gateway to execute the new software.
System is Rebooting ......

Figure 3-81 Step 4 of F/W Upgrading

After upgrade the firmware the device will reboot and execute the new firmware.

3.6.4.2 Config Up/Down

Get or set the whole device configuration by TFTP, FTP or HTTP.

The Access Path: Maintenance>>Device Control>>Config Up/down

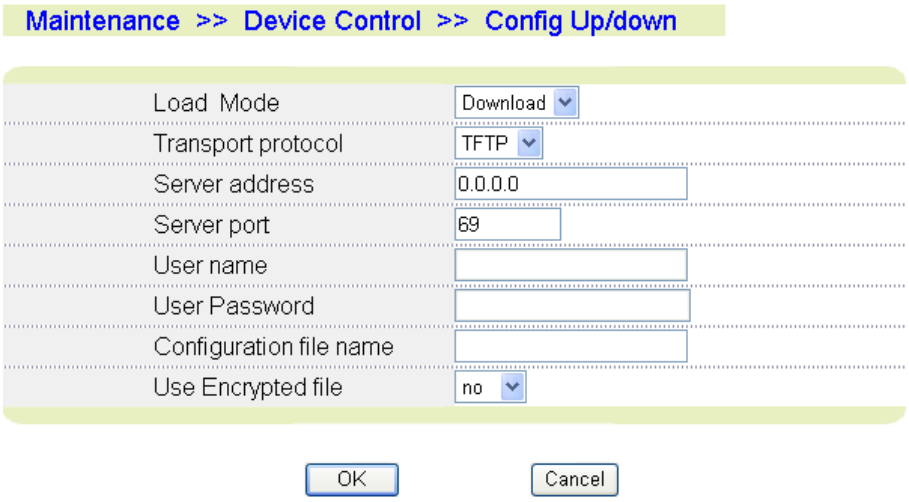


Figure 3-82 Config Up/Down

Table 3-59 Description of Config Up/Down Items

Field	Description
Load Mode	1.Download(default) - Download a configuration file to the device by TFTP. 2.Upload – Upload the configuration of the device to a specify file by TFTP.
Transport protocol	Select a transport protocol
Server Address	Default is “0.0.0.0”.
Server port	Default TFTP Server port is 69.
User name	User name of server
User Password	User password of server
Configuration file name	The configuration File name to be Upload/Download
Use Encrypted file	1.no(default) 2.yes If encrypt transmitting messages of the configuration file uploading or downloading.

Note:

You need to run a TFTP server (such as Cisco TFTP server) for the uploading or downloading except the TFTP server of the server has run.

A. Upload

Step 1: Set uploading parameters.

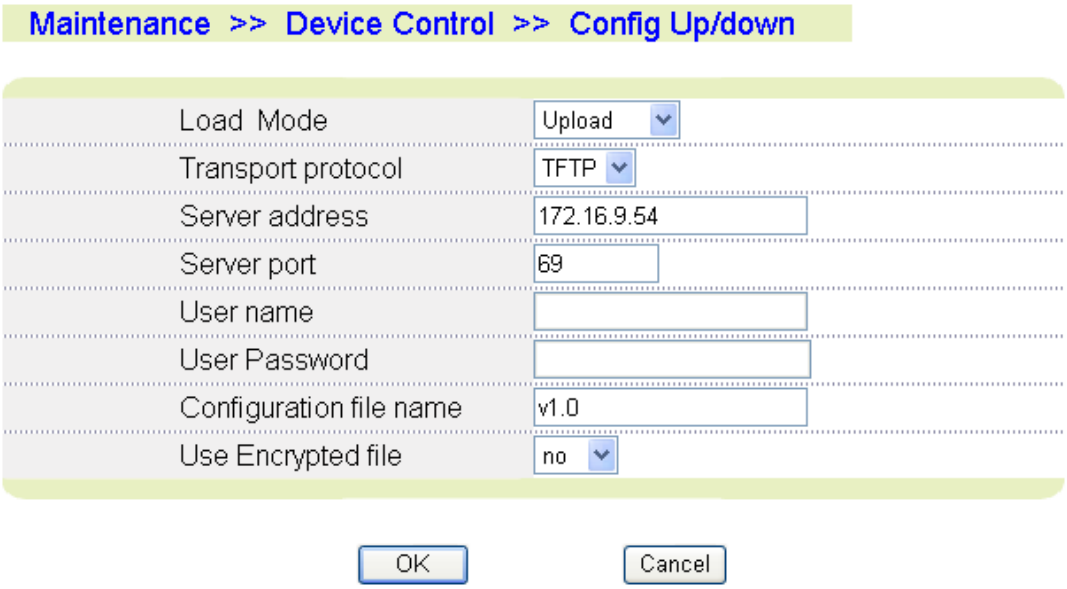


Figure 3-83 Set Config uploading parameters

If the configuration is finished, click “OK” button below the web page to execute uploading or click “Cancel” button to give up. The whole process will take about 5 seconds.

Step 2: Uploading

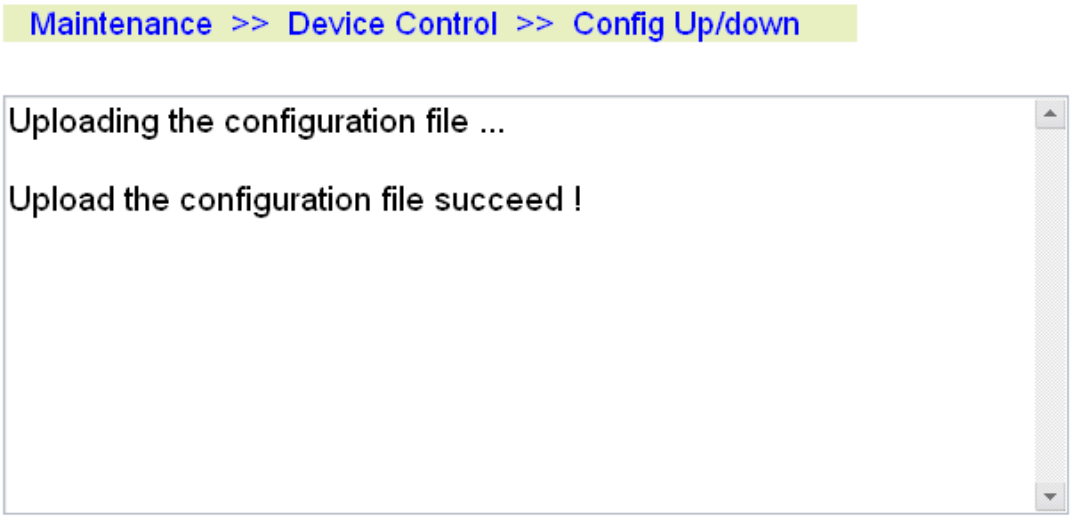


Figure 3-84 Config uploading

If the uploading has succeeded you will find a file named “v1.0” in you TFTP server root directory.

If the TFTP server is not running now, the uploading will abort.

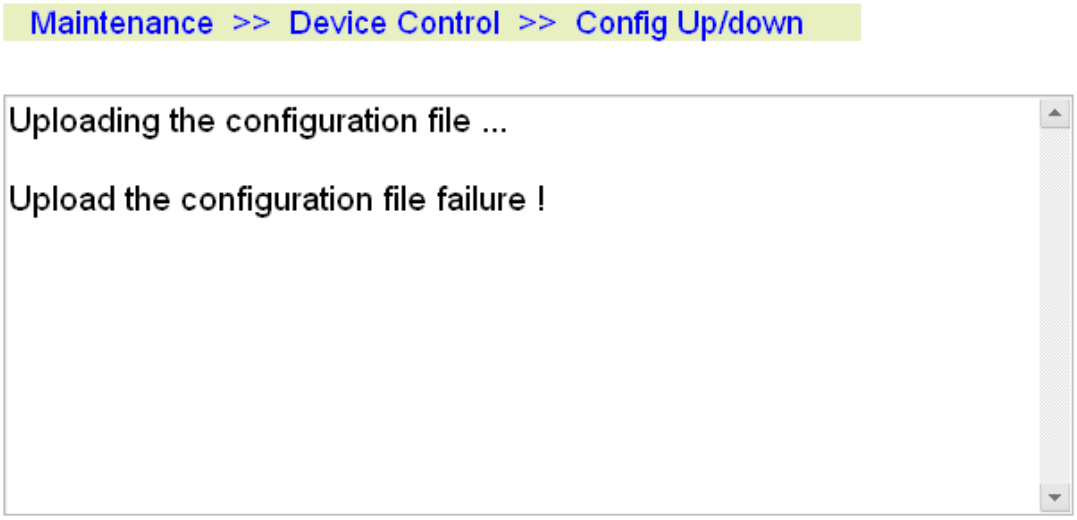


Figure 3-85 Config uploading failure

B. Download

Step 1: Set downloading parameters.

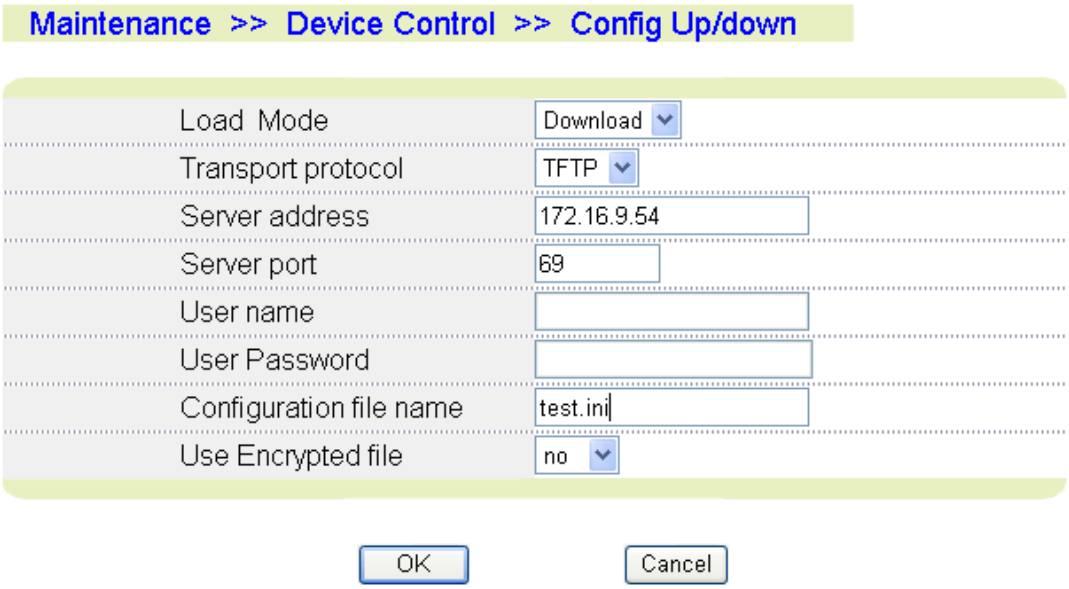


Figure 3-86 Set config downloading parameters

If the configuration is finished, click “OK” button below the web page to execute Upload/Download or click “Cancel” button to give up. The whole process will take about 5 seconds.

Step 2: downloading

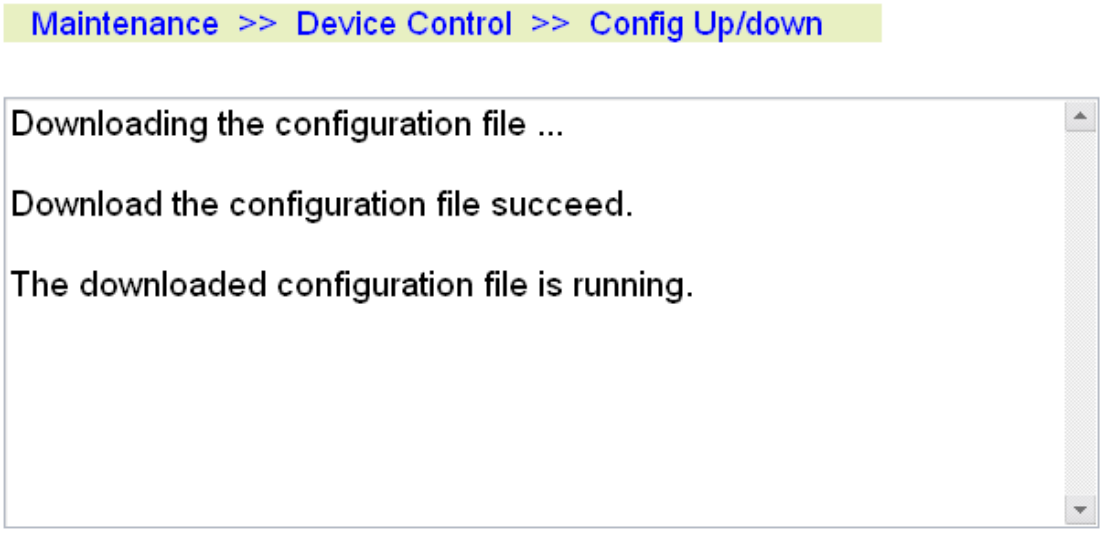

Maintenance >> Device Control >> Config Up/down
Downloading the configuration file ...
Download the configuration file succeed.
The downloaded configuration file is running.

Figure 3-87 Config downloading

If the downloading has succeeded the configuration will be changed to as the config file named “test.ini” in you TFTP server root directory.

If the TFTP server is not running or config file does not exist the downloading will abort.

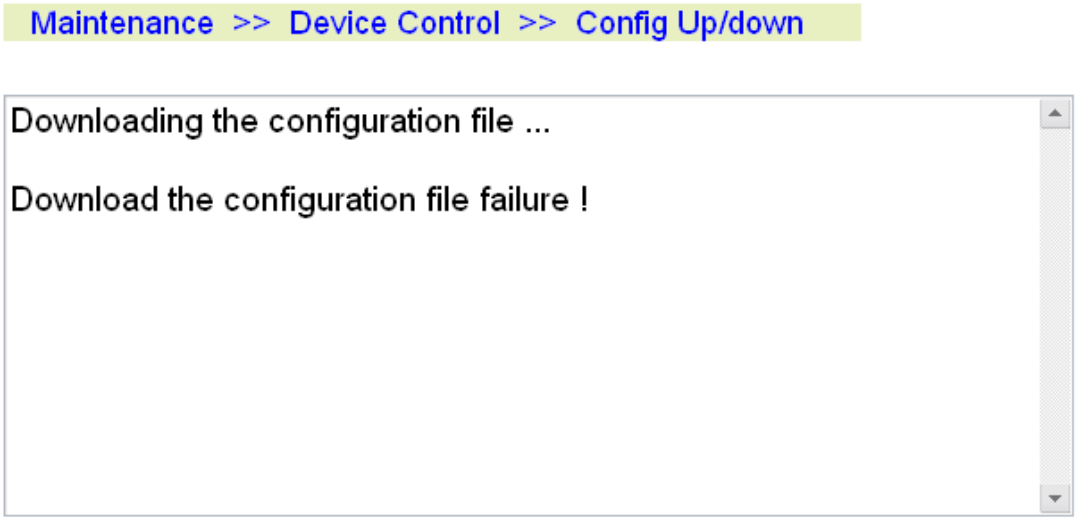

Maintenance >> Device Control >> Config Up/down
Downloading the configuration file ...
Download the configuration file failure !

Figure 3-88 Config downloading failure

3.6.4.3 Syslog Upload

Upload the recorded system log file via TFTP server.

The Access Path: Maintenance>>Device Control>>Syslog Upload

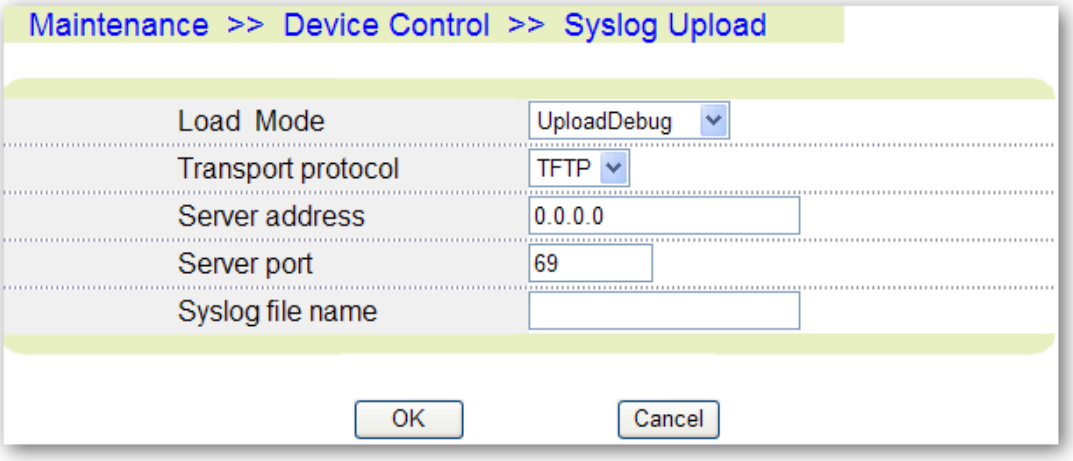


Figure 3-89 Syslog Upload

Table 3-60 Description of Syslog Upload Items

Field	Description
Load Mode	1. Upload – Upload the system log of the device to a specified file by TFTP.
Transport protocol	TFTP protocol
Server address	Default is “0.0.0.0”.
Server port	Default TFTP Server port is 69.
Syslog file name	The system log File name to be Uploaded

3.6.4.4 Reset to Default

This feature is used to reset all the settings to the factory defaults. Selecting this menu will cause a forced reset to the VENUS 29xx Series. The configuration defaults are:

- DHCP Client: Unchanged
- DNS Server: Disable
- Country Code: Generic
- WAN IP Address: Unchanged
- WAN IP Mask: Unchanged
- Default Gateway: Unchanged
- User Name: user
- No Password

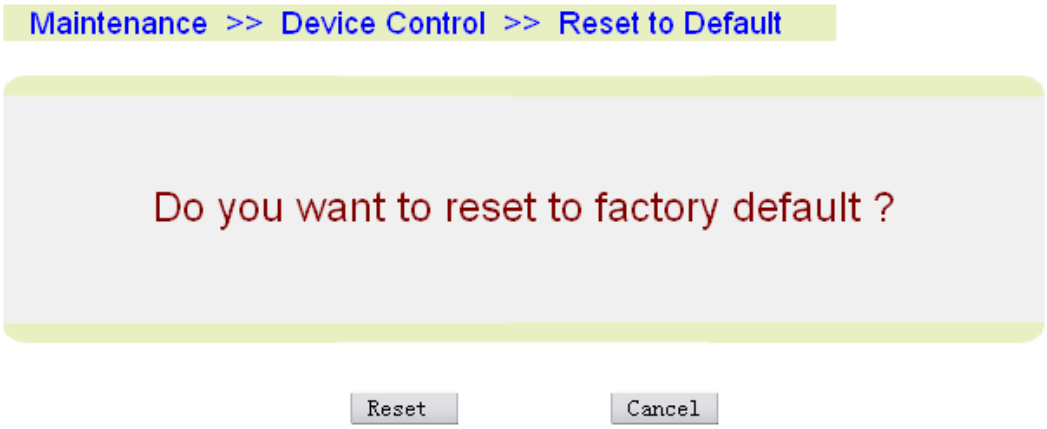


Figure 3-90 Reset to Default

Click the “Reset” button bellow the web page to execute restore or click “Cancel” button to give up.

After click the “Reset” button the device will reboot with the default configuration.

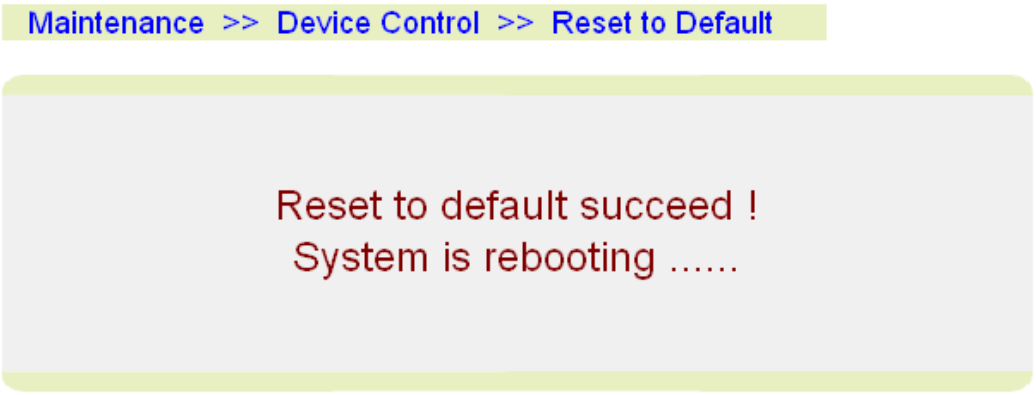


Figure 3-91 Reset finished

Note:

The Ethernet configuration will not be changed in this process.

3.6.4.5 Reboot

After changing some configurations of the device, it may be required to restart the device to activate the new configuration. Use this menu to reset the VENUS 29xx Series.

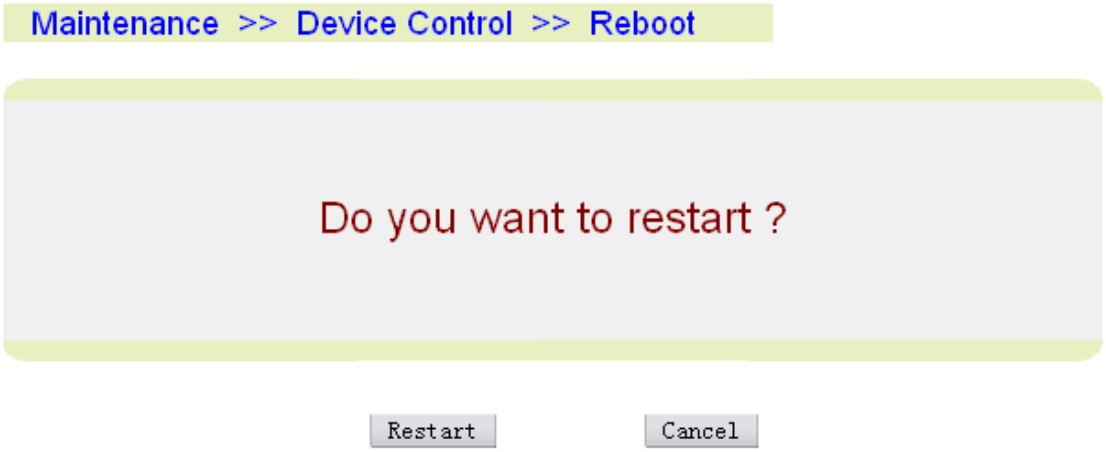


Figure 3-92 Reset to Default

Click the “Restart” button below the web page to reboot or click “Cancel” button to give up.

Caution:

Reset the gateway only when there is no phone call in progress. Otherwise, VoIP service will be abruptly terminated

After click the “Restart” button the device will reboot.

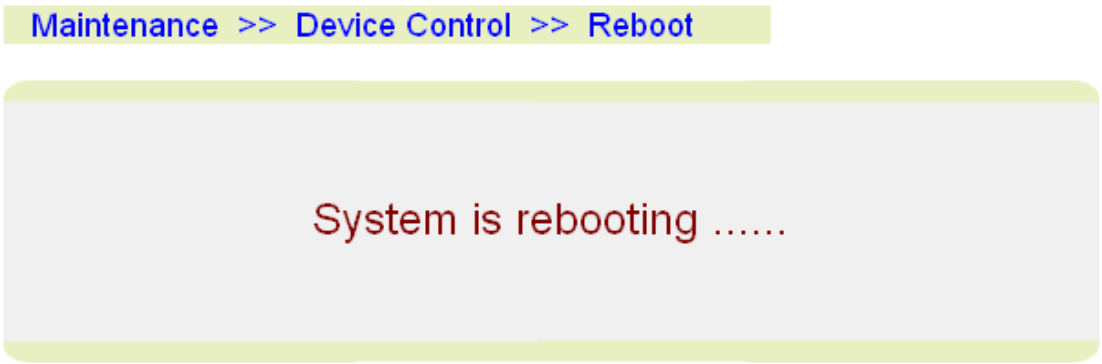


Figure 3-93 Reboot finished

3.7 Status

3.7.1 System information

Click the “System info” of the top menu; come to “System info” page. This Page will display the basic system information of the VENUS 29xx as in the following figure.

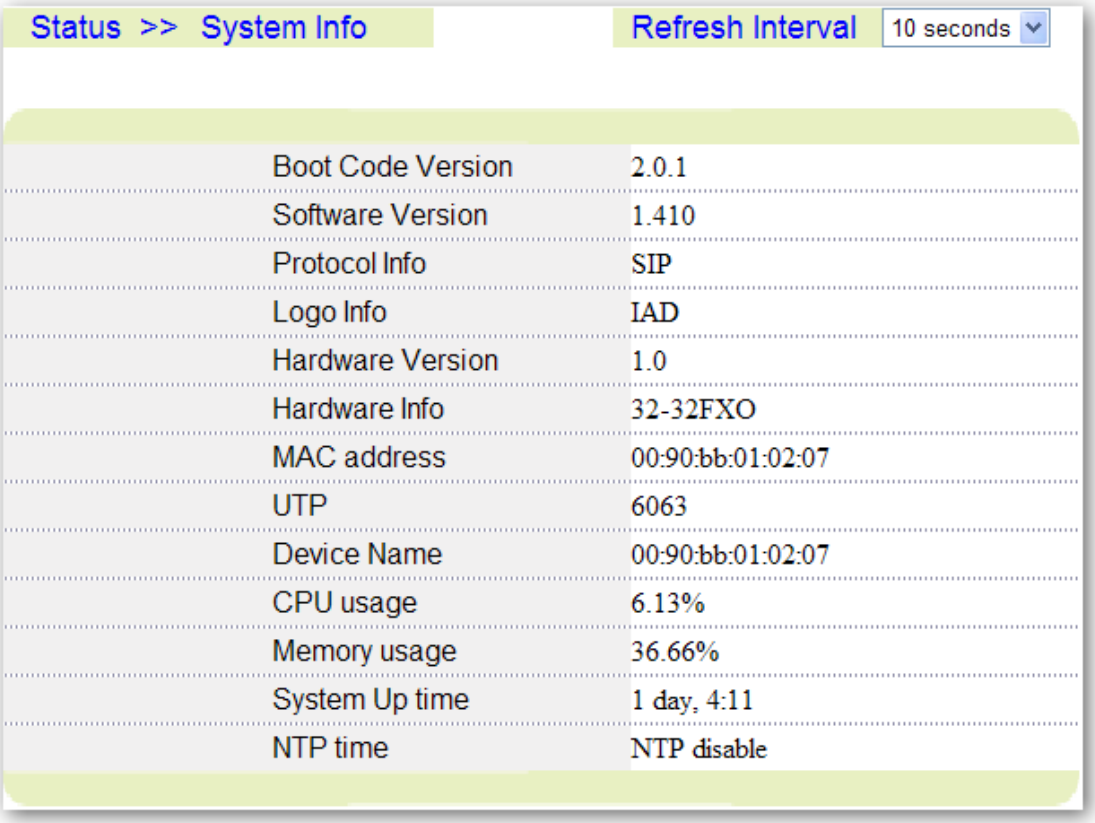


Figure 3-94 System information Status

3.7.2 Ethernet

Click the “Ethernet” of the top menu; come to “Ethernet” page. This Page will display Ethernet information of the VENUS 29xx in the following figure.

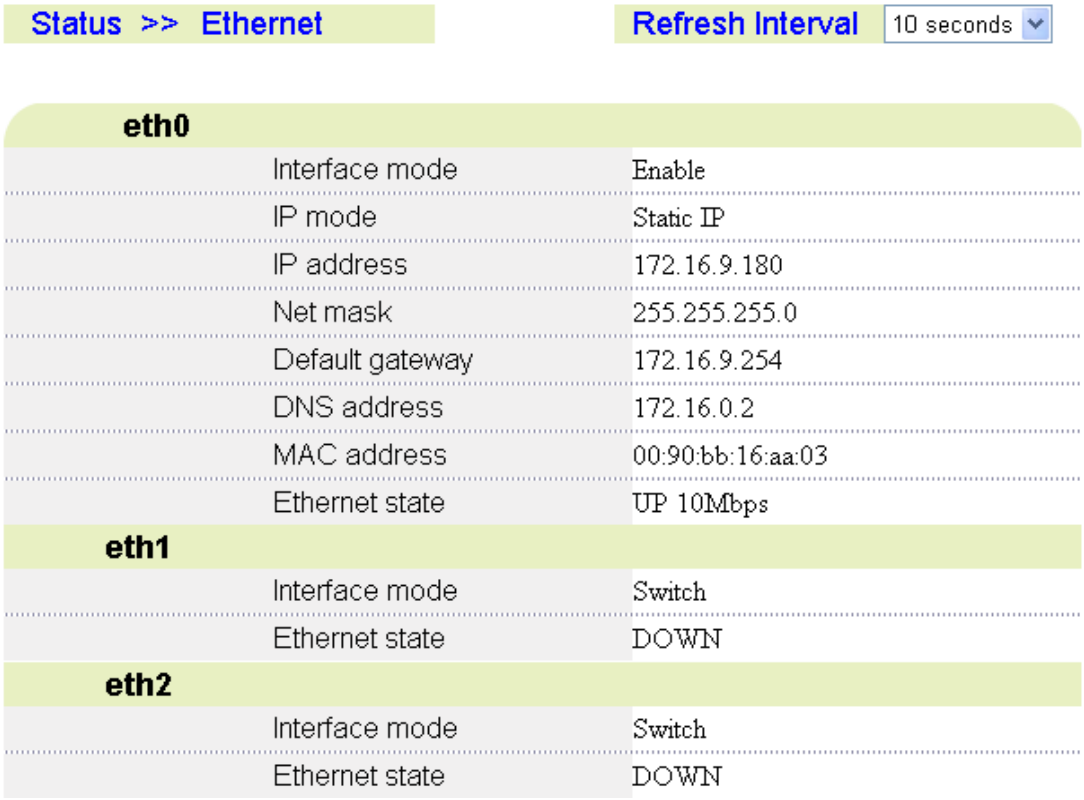
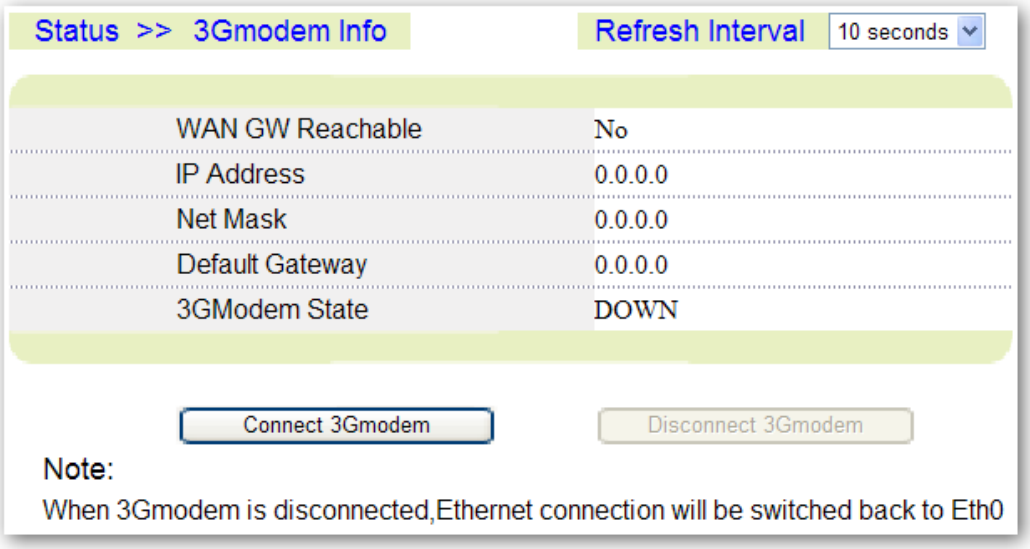


Figure 3-95 Ethernet Status

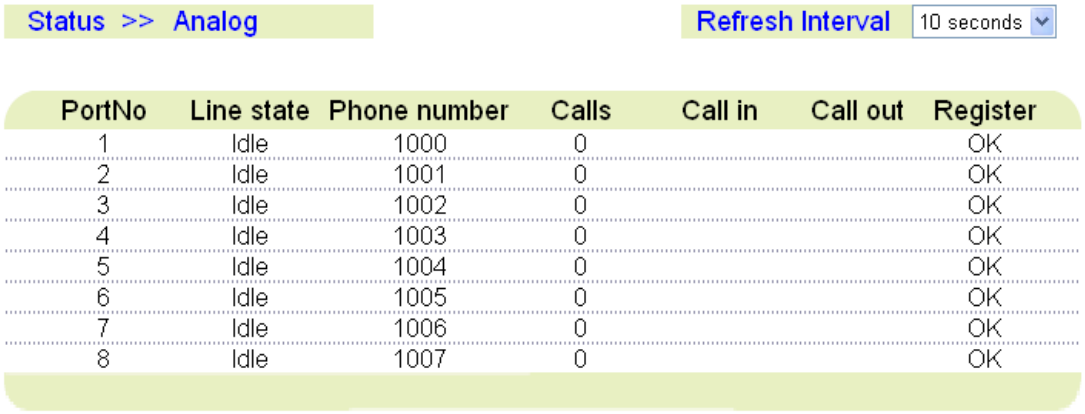
3.7.3 3Gmodem Info

3Gmodem Info shows the current status of 3G modem connection, user can also select to connect or disconnect 3G modem connections.



3.7.4 Analog (FXS/FXO)

Click the “Analog (FXS/FXO)” of the top menu; come to “Analog” page. This Page will display Analog information of the VENUS 29xx in the following figure.



PortNo	Line state	Phone number	Calls	Call in	Call out	Register
1	Idle	1000	0			OK
2	Idle	1001	0			OK
3	Idle	1002	0			OK
4	Idle	1003	0			OK
5	Idle	1004	0			OK
6	Idle	1005	0			OK
7	Idle	1006	0			OK
8	Idle	1007	0			OK

Figure 3-96 Analog Status

3.7.5 Routing Table

Click the “Routing Table” of the top menu; come to “Routing Table” page. This Page will display Routing Table information of the Proxy Server in the following figure.

Destination	Gateway	Netmask	Flags	Metric	Ref	Use	Iface
192.168.1.0	0.0.0.0	255.255.255.0	U	0	0	0	eth0
127.0.0.0	0.0.0.0	255.0.0.0	U	0	0	0	lo
0.0.0.0	192.168.1.20	0.0.0.0	UG	0	0	0	eth0

Figure 3-97 Routing Table Status

3.7.6 ARP Table

Click the “ARP Table” of the top menu; come to “ARP Table” page. This Page will display ARP Address information of the Proxy Server in the following figure.

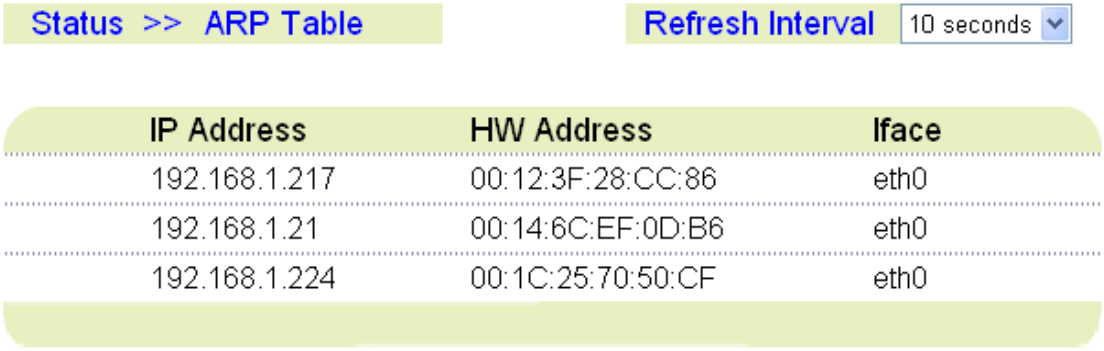
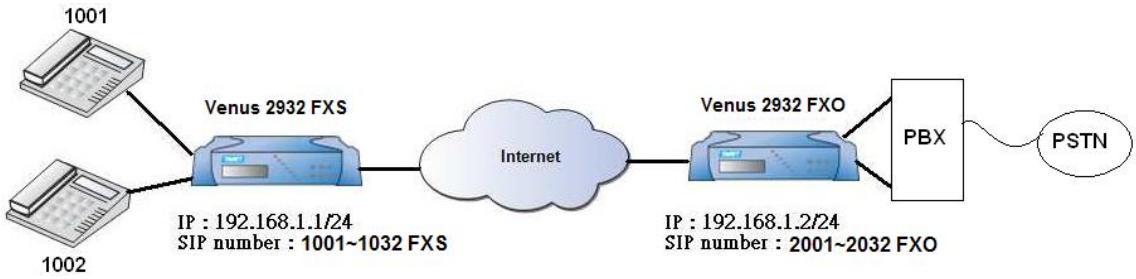


Figure 3-98 ARP Table Status

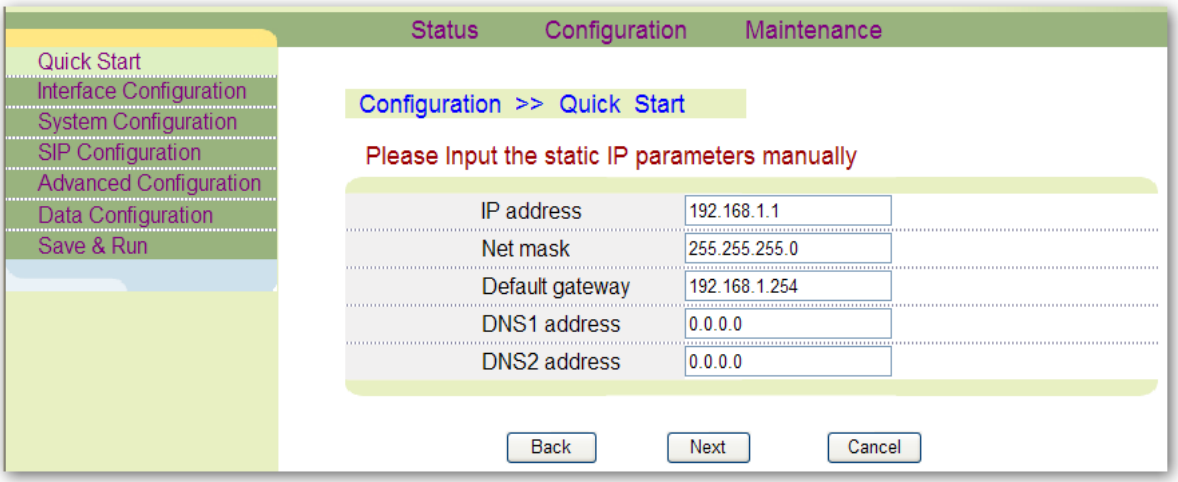
Table 3-61 Description of Routing Table Monitoring Items

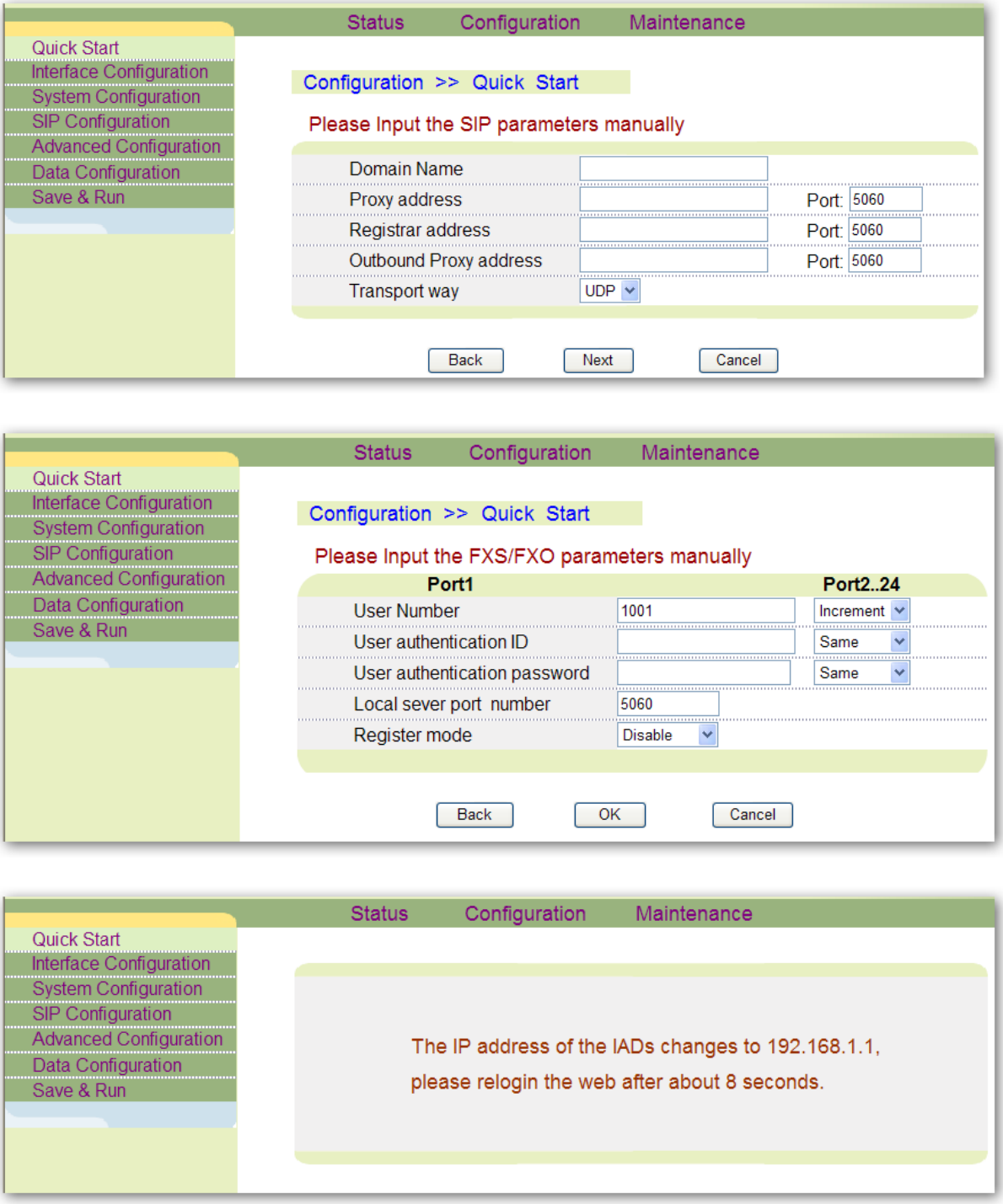
Field	Description
IP Address	Proxy Server IP Address
HW Address	Proxy Serve MAC Address
Iface	Interface Eth0, Eth1 or Eth2~4

3.8 FXS to FXO line extension example

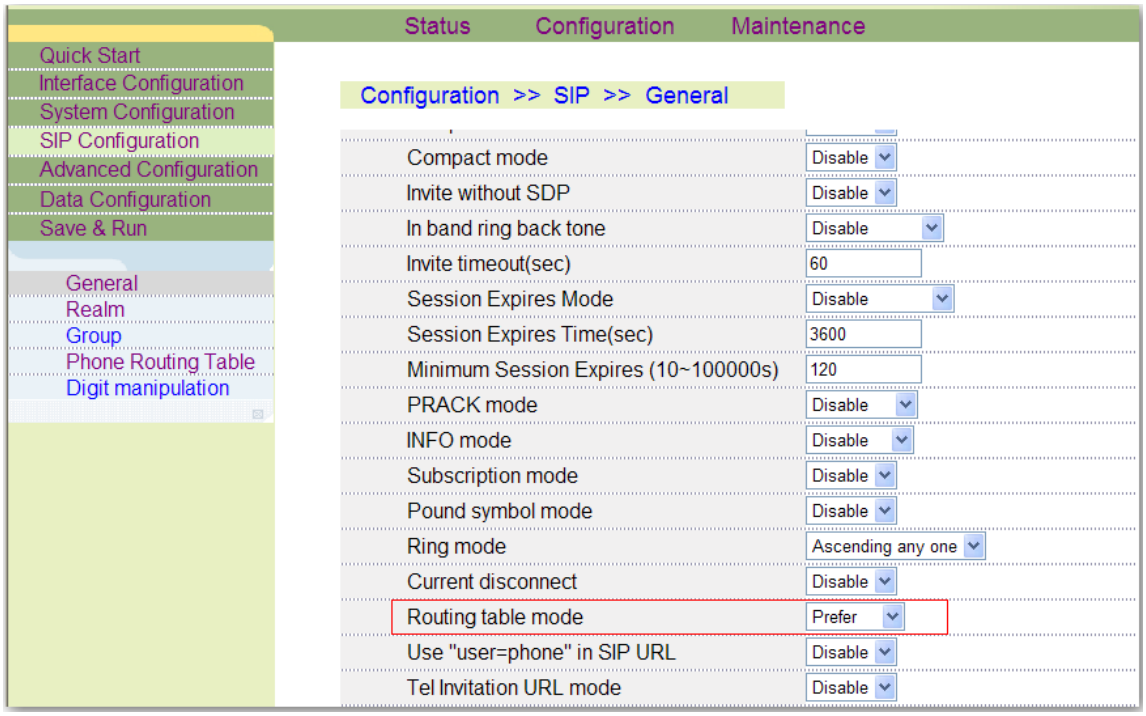


(1). Venus 2932 FXS : Configuration -> Quick Start -> Static IP





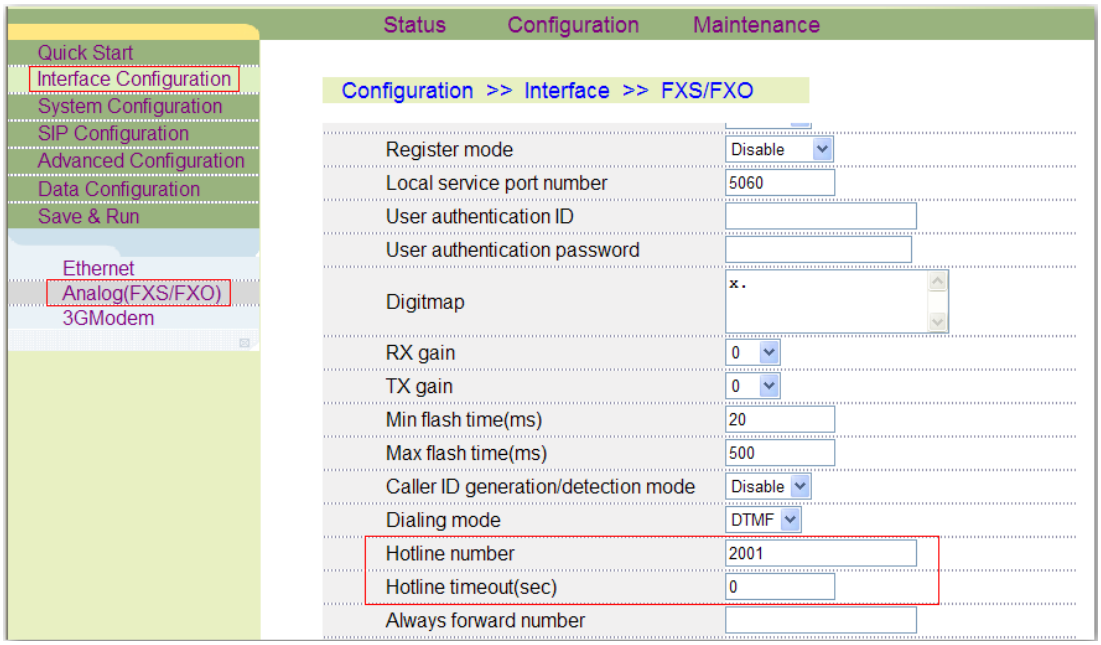
(2). Venus 2932 FXS : Configuration -> SIP Configuration -> General



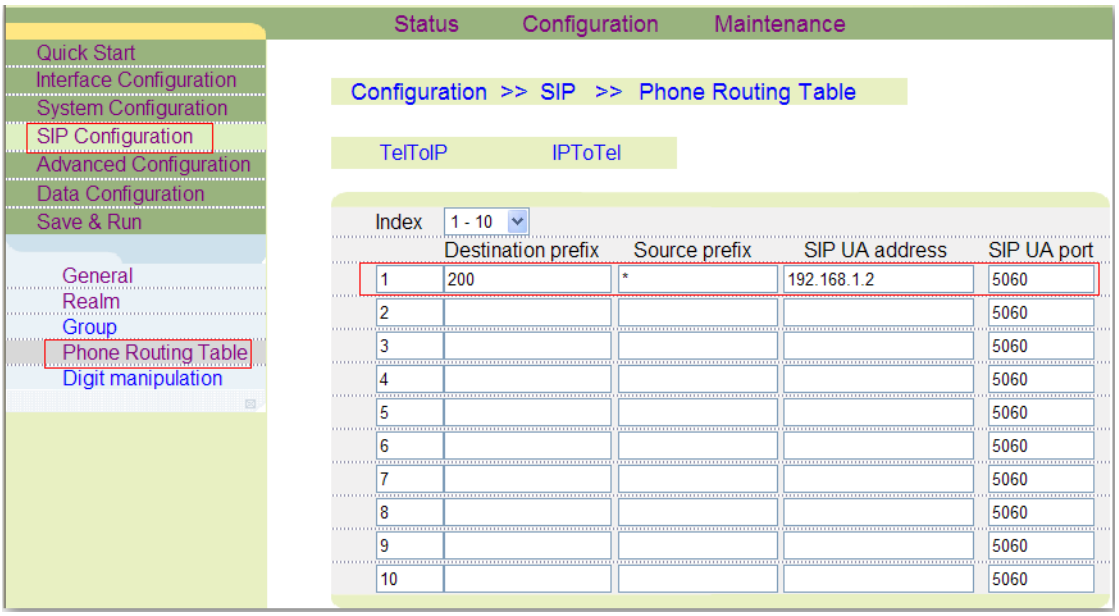
(3). Venus 2932 FXS : Configuration -> Interface Configuration -> Analog(FXS/FXO)

Configure hotline as 2001 for Port 1 (1001) since this FXS port will dial out to remote FXO directly.

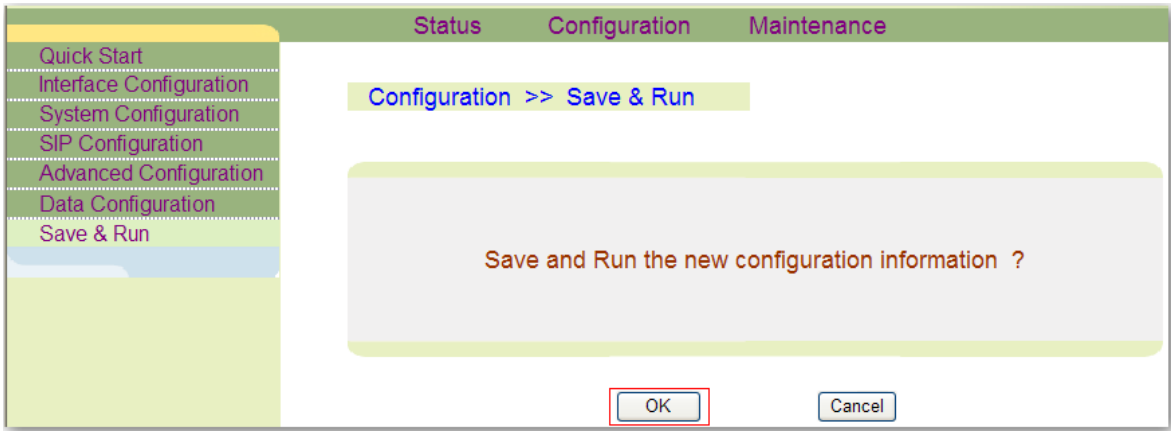
Configure hotline as 2002 for Port 2 (1002) since this FXS port will dial out to remote FXO directly.



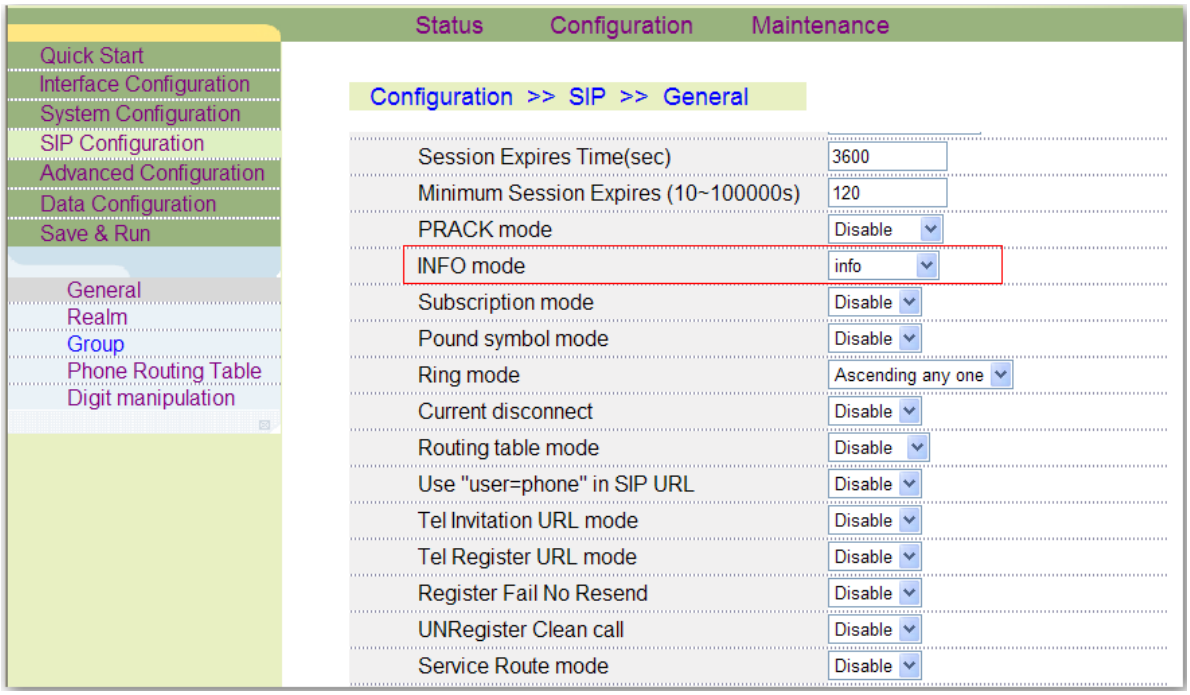
(4). Venus 2932 FXS : Configuration -> SIP Configuration -> Phone Routing Table



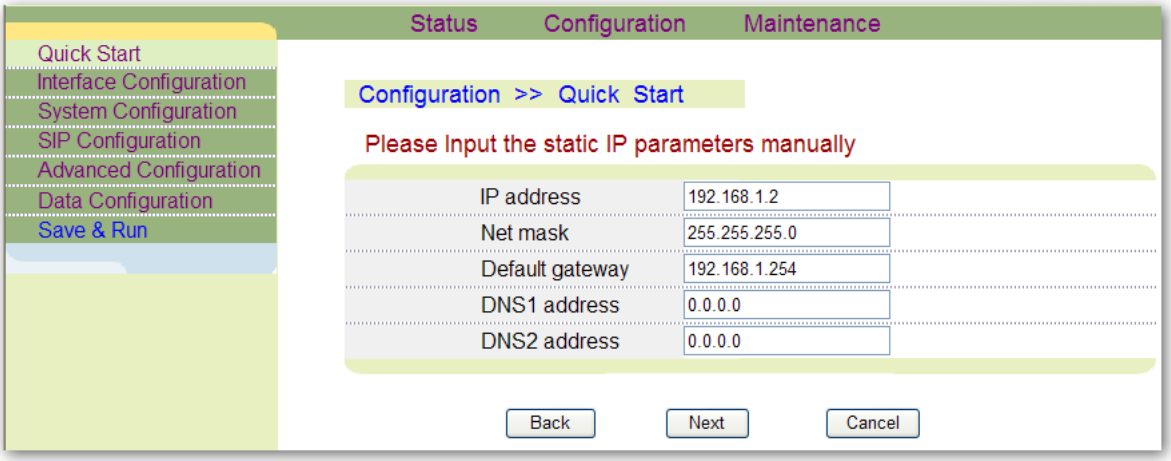
(5). Venus 2932 FXS : Configuration -> Save & Run

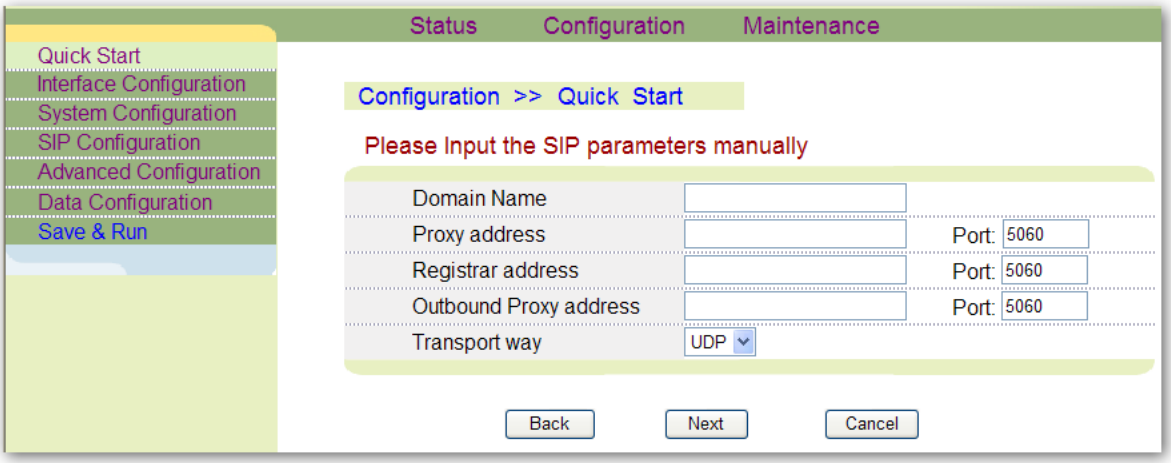


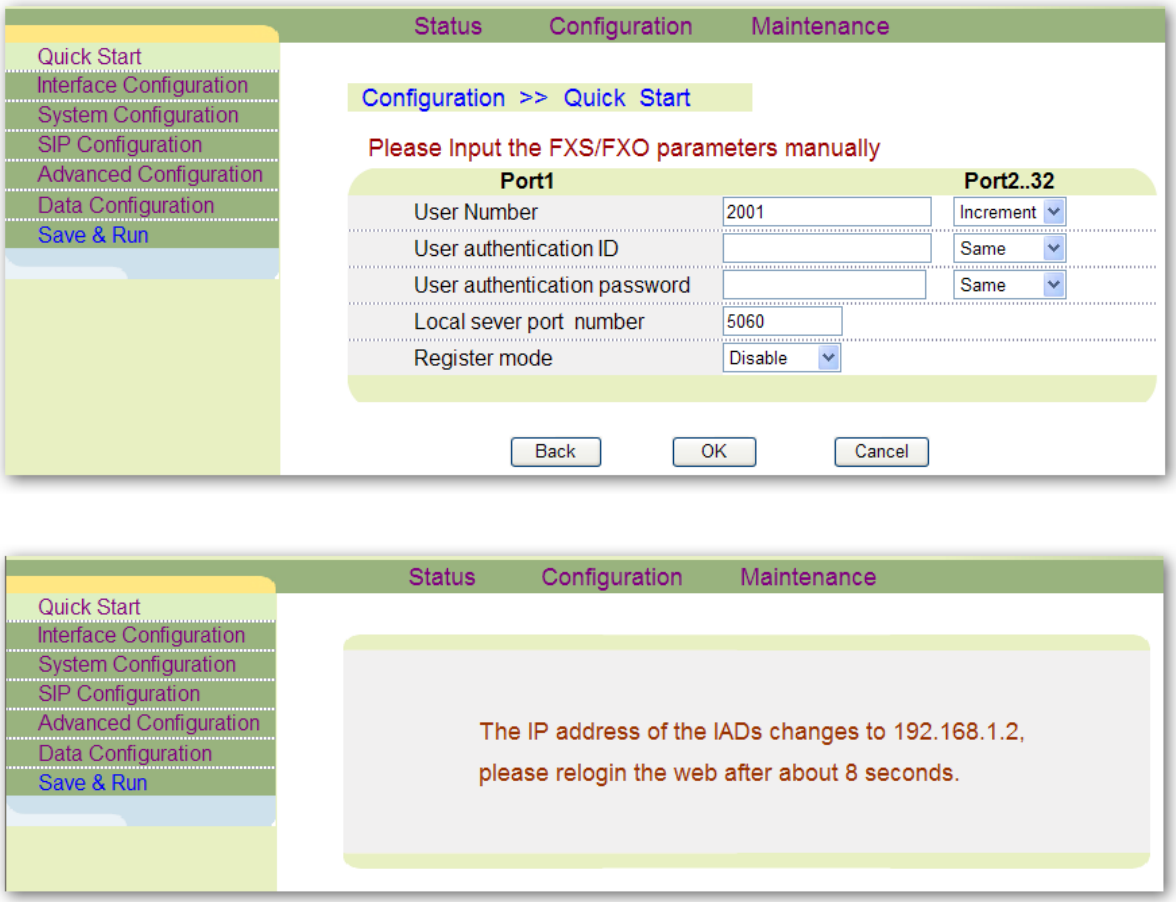
(6). Venus 2932 FXS : Configuration -> SIP Configuration-> General-> INFO mode-> info



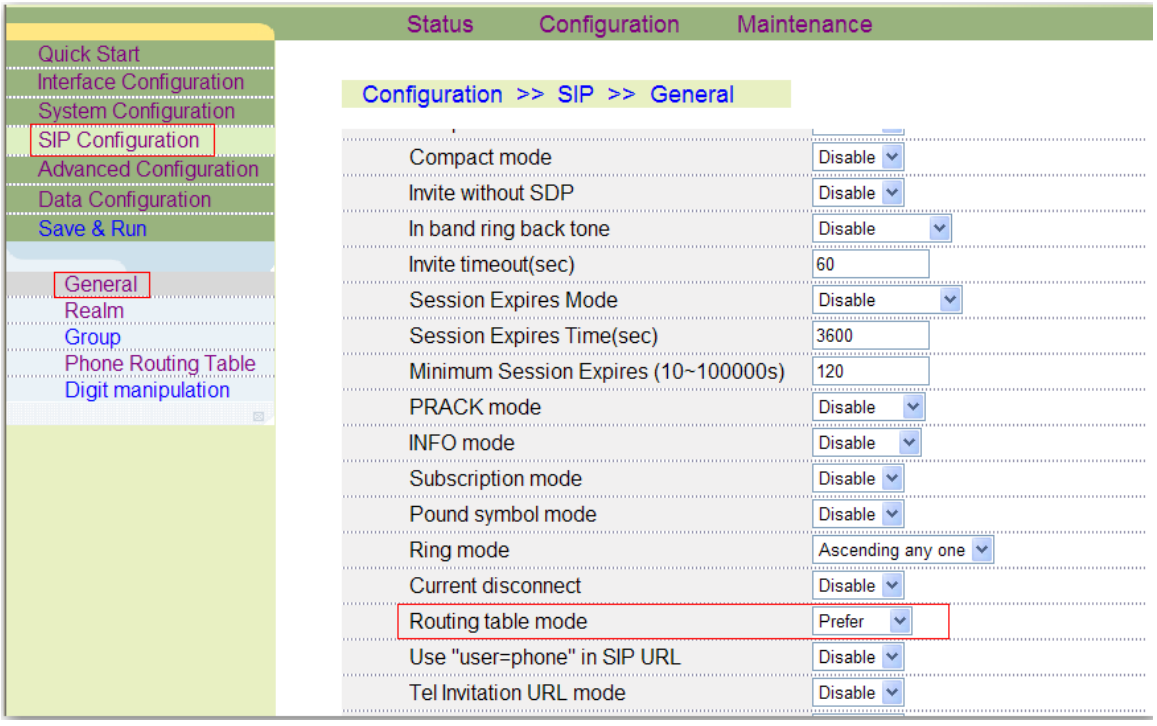
(7) Venus 2932 FXO : Configuration -> Quick Start -> Static IP



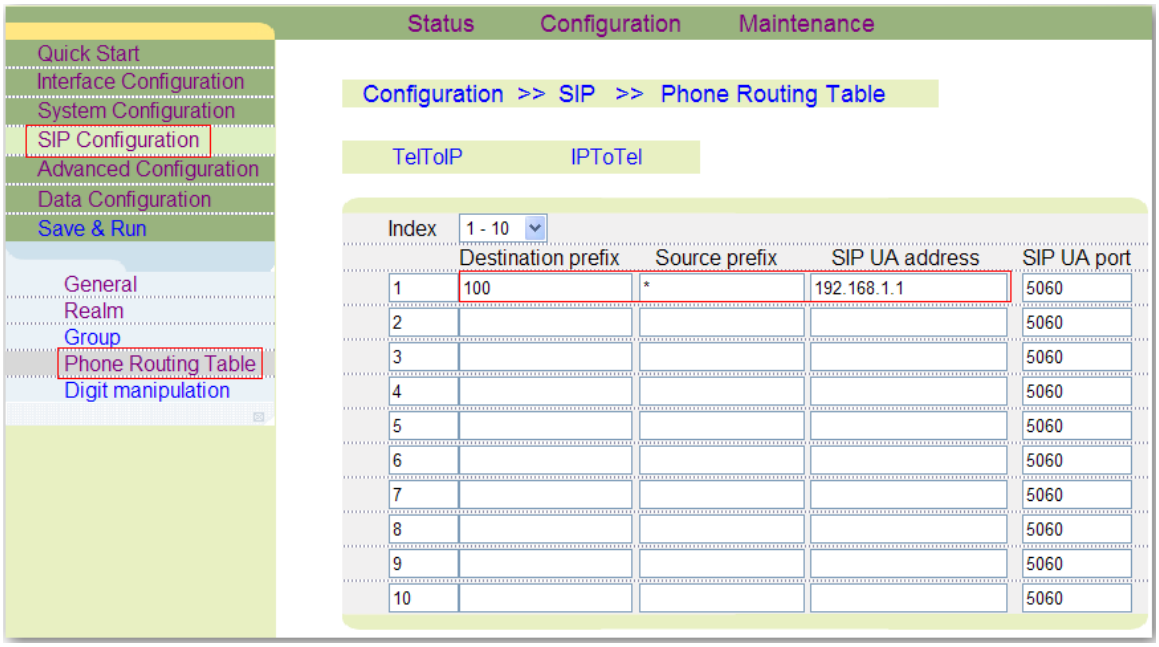




(8). Venus 2932 FXO : Configuration -> SIP Configuration -> General

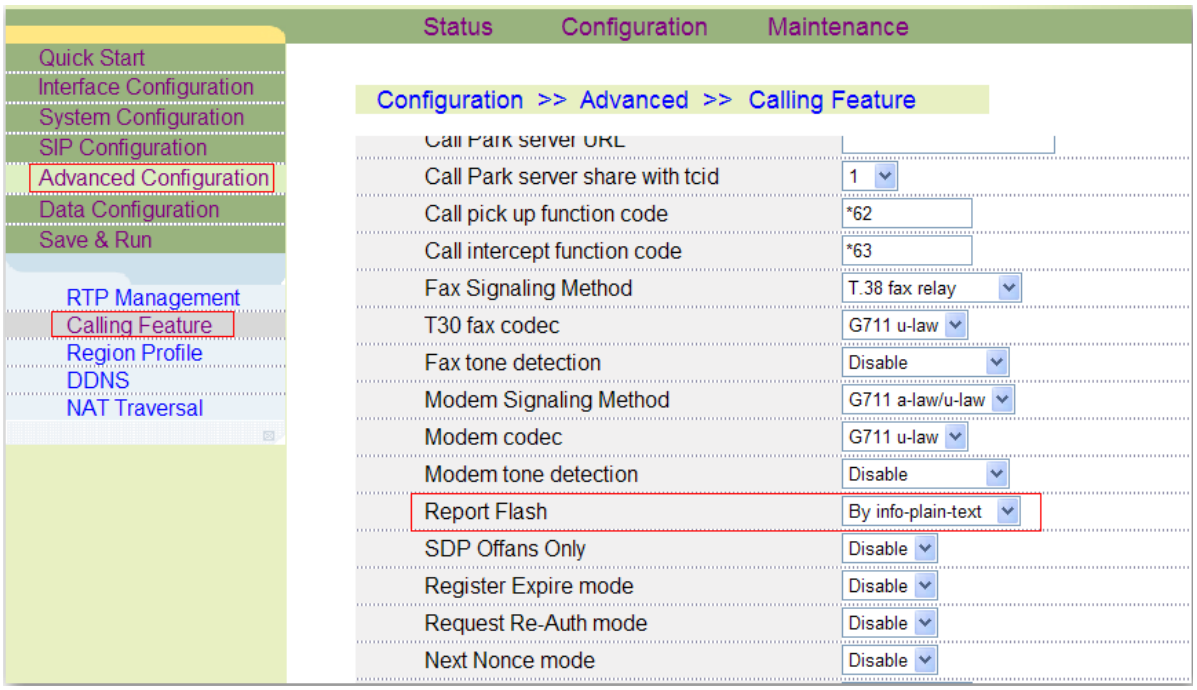


(9). Venus 2932 FXO : Configuration -> SIP Configuration -> Phone Routing Table



(10). Venus 2932 FXO : Configuration -> Save & Run

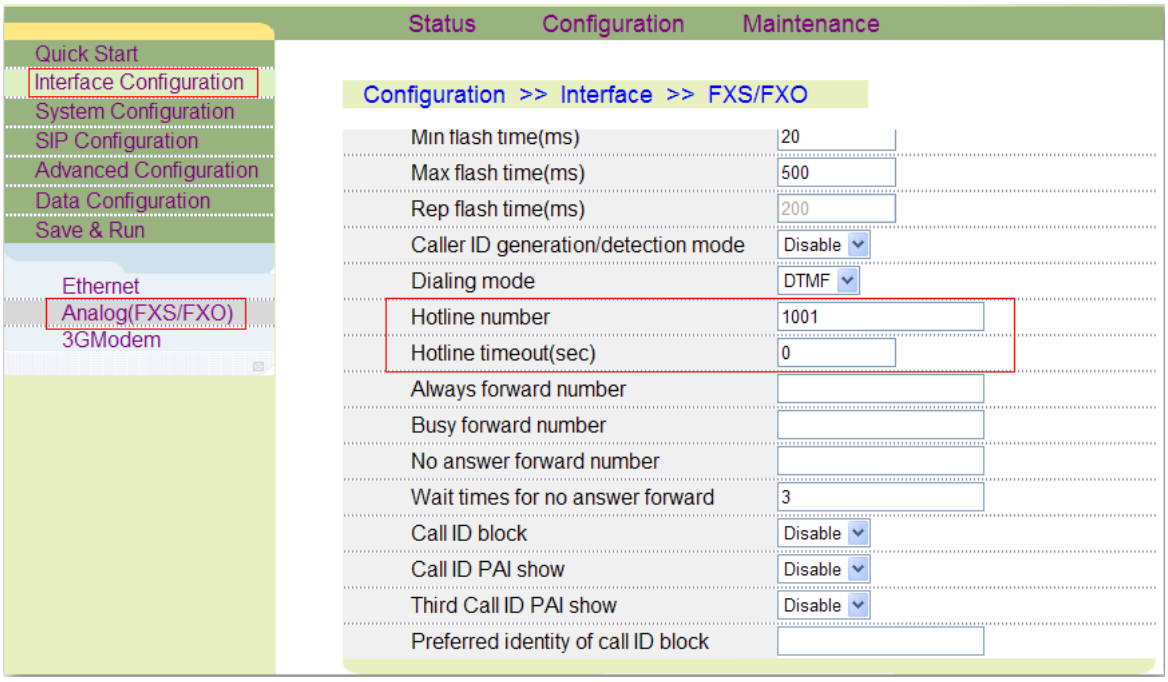
(11). Venus 2932 FXO : Configuration -> Advanced Configuration -> Calling Feature -> Report Flash -> By info-plain-text



(12). Venus 2932 FXO : Configuration -> Interface Configuration -> Analog(FXS/FXO)

Configure hotline as 1001 for Port 1 (2001) since this FXO port will dial out to remote FXS directly

Configure hotline as 1002 for Port 2 (2002) since this FXO port will dial out to remote FXS directly



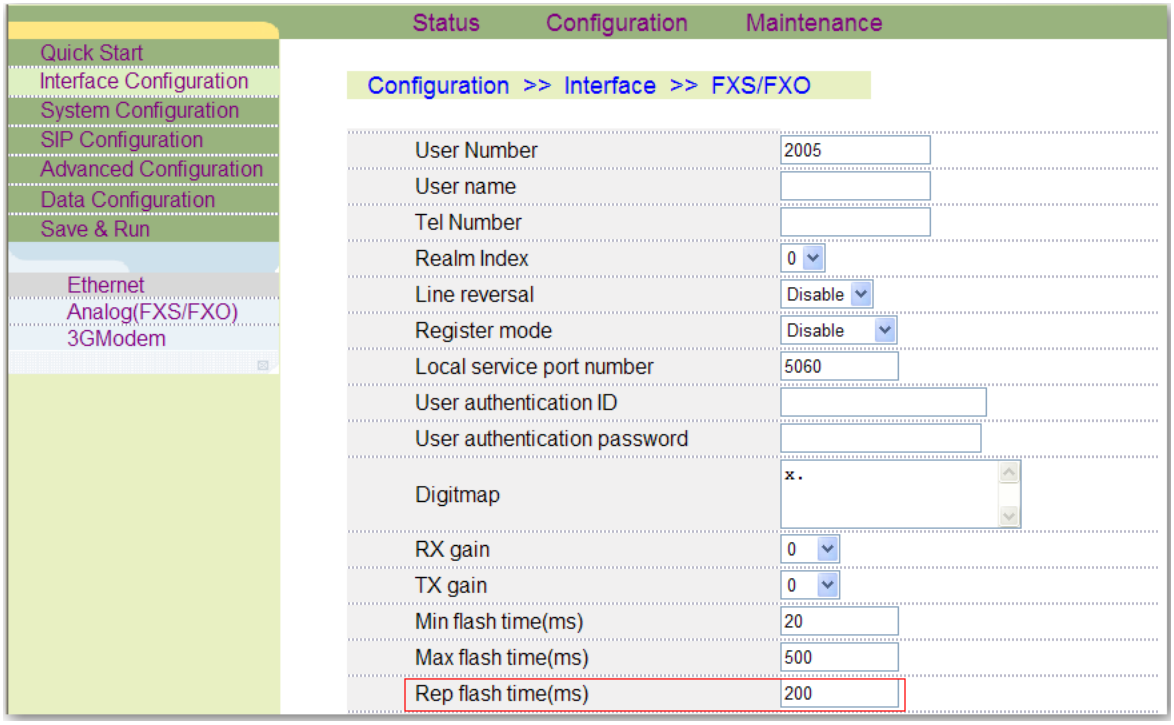
(13). Venus 2932 FXO : Configuration -> Interface Configuration ->

Analog(FXS/FXO)

→ Rep flash time(ms) default value is 200

This option is used for controlling the flash timer which FXO will generate when it receives flash notice from FXS INFO message.

Depends on the PBX or PSTN line connects to this FXO port, the value can be adjusted for better compatibility.



When setup complete, when user on FXS port off-hooks, it will be directly routed to remote FXO port. When FXO port is being called from PSTN or PBX side, the call will be directed to remote FXS port.

3.9 Navigation the Command Line Interface

The Command Line Interface is designed mostly for technical support engineers. It provides a Command Line Interface (CLI) where user can enter commands

- To setup detailed configuration
- To monitor detailed system status messages, and
- To diagnose the device

```
**********************************************************************
Sip version 1.00 compiled on Jun  3 2008 at 12:40:33
IAD 2908-8FXS  Hardware version 1.0
**********************************************************************
Login:admin
Password:
type help for more infomation.
admin>_
```

Figure 3-99 CLI Main UI

Within the CLI, enter “?” or “help”, the device will display all the commands available.

```
admin>help
MXP Debug Application Command List
                Command Description
                      ? Displays Command Help Table
                   help Displays Command Help Table
---------------------------- User Commands
                    spy Set/Display Spy levels
                     st Status Commands
                   show Show System Configuration
                    set Set System Configuration
                     mt Maintenance Commands
                   save Save System Configuration
                 logout Logout System
```

Figure 3-100 Main CLI Tree

Table 3-62 The Main CLI Tree

Type	Sub-Type	Description
Sys Commands	?	Displays Command Help Table
	help	Displays Command Help Table
Status Commands	st	Displays System Status Information
Configuration Commands	save	Save System Configuration
	set	Set System Configuration
	show	Show System Configuration
Maintenance Commands	mt	Maintenance or Diagnostics the system
Other Commands	spy	Set the spy level for debug
	logout	Logout the system

Caution:

Activating some of the commands supported in the Command Line Interface (CLI) during run-time may affect the normal service of the device.

3.9.1 Syntax

The command line syntax is “command parameter …”.

The following are some example commands:

set tcid 3 rxgain –5

venus card 4 fxs ring_freq 20

3.9.2 Status Commands

Display the system status information.

```
admin>st
st Commands:
  st si              : System information
  st g3info          : 3gModem state information
  st eth             : Ethernet state information
  st analog          : Analog state information
  st rt              : Route table information
  st arp             : ARP table information
OK
```

Figure 3-101 Status Commands

Table 3-63 st Command Items

Command	Description
st si	System information
st g3info	3G modem information
st eth	Ethernet information
st analog	Analog information
st rt	Route table information
st arp	ARP table information

3.9.3 Configuration Commands

3.9.3.1 Save

Save the parameters have been changed to system.

```
admin>save

Saving the configuration, please wait ...
The information has been saved and is running!
admin>_
```

Figure 3-102 Prompt of Save command

Table 3-64 Save Command

Command	Description
Save	Save the configuration of the VENUS 29xx

3.9.3.2 Set

Modify system parameters.

```
admin>set
Use following for more info
  set interface <options>
  set system <options>
  set sip <options>
  set advance <options>
  set data <options>
  set syslogcf <options>
OK
```

Figure 3-103 Set command

Table 3-65 Set Command Items

Command	Sub-Command
Set interface <options> (admin > set interface)	set interface ethernet <options>
	set interface analog <options>
	set interface 3gmodem <options>
	set interface pstn<options>
Set system <options> (admin > set system)	set system general <options>
	set system ntp <options>
	set system ui <options>
	set system security <options>
	set system provision <options>
	set system snmp <options>
	set system tr069 <options>
	set system threshold <options>
Set sip <options> (admin > set sip)	set sip general <options>
	set sip realm <options>
	set sip group <options>
	set sip phone_route <options>
	set sip call_filter <options>
	set sip mainp_tel2ip_dst <options>
	set sip mainp_tel2ip_src <options>
	set sip mainp_ip2tel_dst <options>
	set sip mainp_ip2tel_src <options>
Set advanced <options> (admin > set advance)	set advance rtp <options>
	set advance call_feature <options>
	set advance region <options>
	set advance nat_traversal <options>
	set advance ddns <options>
	set advance dsp <options>
	set advance coding <options>

Command	Sub-Command
	set advance tiu <options>
Set data <options> (admin > set data)	set data dhcp <options>
	set data nat_mapping <options>
	set data static_route <options>
	set data qos <options>
	set data vlan<options>

3.9.3.3 Show

Show system parameters and state.

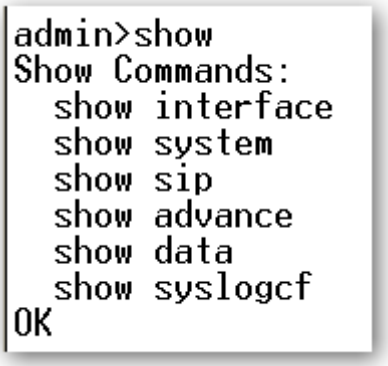


Figure 3-104 Show command

Table 3-66 Show Command Items

Command	Sub-Command	Description
Show interface (admin > show interface)	show interface ethernet <index|all>	Display Ethernet port information
	show interface analog <index|all>	Display voice port information
	show interface 3gmodem	Display 3G modem information
	show interface pstn	Display PSTN port information
Show system (admin > show system)	show system general	Display general system information
	show system ntp	Display NTP settings information
	show system ui	Display User Interface settings information
	show system security	Display security settings information
	show system provision	Display provision settings information
	show system snmp	Display SNMP settings information
	show system tr069	Display TR069 settings information
	show system threshold	Display threshold settings information
Show sip (admin > show sip)	show sip general	Display general SIP settings information
	show sip realm <index|all>	Display SIP realm settings information
	show sip group <index|all>	Display SIP group settings information
	show sip phone_route <index|all>	Display SIP phone route settings information
	show sip call_filter <index|all>	Display SIP call filter settings information
	show sip mainp_tel2ip_dst <index|all>	Display SIP Digit Manipulation tel to IP destination settings information.

	show sip mainp_tel2ip_src <index|all>	Display SIP Digit Manipulation tel to IP source settings information.
	show sip mainp_ip2tel_dst <index|all>	Display SIP Digit Manipulation IP to tel destination settings information.
	show sip mainp_ip2tel_src <index|all>	Display SIP Digit Manipulation IP to tel source settings information.
Show advanced (admin > show advance)	show advance rtp	Display advanced RTP settings information
	show advance call_feature	Display call feature settings information
	show advance region	Display advanced region settings information
	show advance nat_traversal	Display advanced NAT traversal settings information
	show advance ddns	Display advanced DDNS settings information
	show advance dsp	Display advanced DSP settings information
	show advance coding <index|all>	Display advanced coding settings information
	show advance tiu	Display advanced tiu settings information
Show data (admin > show data)	show data dhcp	Display DHCP settings information
	show data nat_mapping	Display NAT mapping settings information
	show data static_route	Display static route settings information
	show data qos	Display QoS settings information
	show data vlan	Display VLAN settings information

3.9.4 Maintenance Commands

Maintenance and diagnostics of the system.

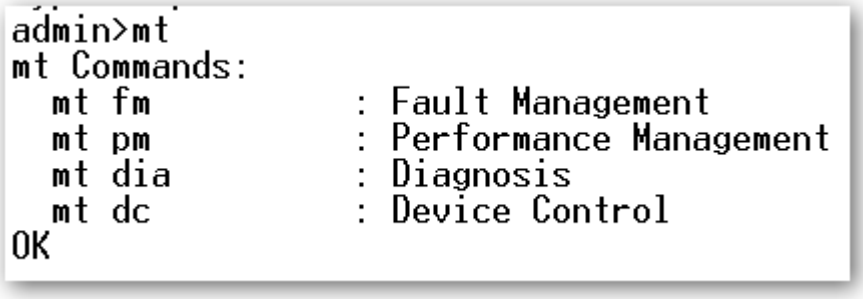


Figure 3-105 Maintenance commands

Table 3-67 mt Command Items

Command	Sub-Command	Description
	mt fm monitor	Show the fm records
	mt fm clear	Clear the fm records
mt pm	Mt pm top	Show the top information of applications
	mt pm eth	Show the ethernet interface information
	mt pm rtp	Show the RTP statistic information
	mt pm cc	Show the call statistic information
	mt pm agent	Show the sip statistic information
	mt pm ci	Show the call initial statistic information
	mt pm nat	Show the NAT statistic information
	mt pm dhcps	Show the DHCP Server statistic information
	mt pm clear	Clear the statistic information
mt dia	mt dia stun	stun information
	mt dia ping	Ping information
	mt dia traceroute	Trace route information
	mt dia lt	line test
	mt dia ps	Process information
	mt dia ls	List file information
	mt dia cat	File cat command
	mt dia kill	Kill command
	mt dia memory	Memory information
	mt dia ccu	Control Coordination Unit

	mt dia st	Signaling trace
	mt dia tr069	TR069 information
	mt dia sip	SIP information
	mt dia mii	Test mii register
	mt dia ilt	Inter line test
	mt dia da	Dsp alloction
mt dc	mt dc upgrade	Firmware upgrade
	mt dc download	Configure file download
	mt dc upload	Configure file upload
	mt dc restore	Configuration information Restore to default
	mt dc reboot	Device warm start

3.9.5 Other Commands

Spy

Open the spy as specify level. The spy messages are used to trace the running of the system.

```
spySPY Debug Commands:
  spy           -- Show table
  spy key       -- Show level of key specfied
  spy key level -- Set level
  spy -l        -- Show possible levels
  spy default   -- Set levels to default
-----  ---------------  ----------------------  -----------
       Key              Level                   Destination
-----  ---------------  ----------------------  -----------
(  1) ROOT             Minor Unexpected Event   dbgPort
(  2) SYSLOG           General Information      dbgPort
(  3) SHMEM            Minor Unexpected Event   dbgPort
(  4) NWIF             Minor Unexpected Event   dbgPort
(  5) ISU              Minor Unexpected Event   dbgPort
(  6) ISUCPDET         Minor Unexpected Event   dbgPort
(  7) CCU              Minor Unexpected Event   dbgPort
(  8) CCUTONE          Minor Unexpected Event   dbgPort
(  9) CCUCPDET         Minor Unexpected Event   dbgPort
( 10) RTCP             Minor Unexpected Event   dbgPort
( 11) DSU              Minor Unexpected Event   dbgPort
( 12) RPU              Minor Unexpected Event   dbgPort
( 13) DEX              Minor Unexpected Event   dbgPort
( 14) DIM              Minor Unexpected Event   dbgPort
( 15) DIM_DNLD         Minor Unexpected Event   dbgPort
( 16) TIU              Minor Unexpected Event   dbgPort
( 17) TIUPDR           Minor Unexpected Event   dbgPort
( 18) DSPA             Minor Unexpected Event   dbgPort
( 19) DCM              Minor Unexpected Event   dbgPort
( 20) CPMU             Minor Unexpected Event   dbgPort
( 21) XSPU             Minor Unexpected Event   dbgPort
( 22) MSUIM            Minor Unexpected Event   dbgPort
( 23) NMM              Minor Unexpected Event   dbgPort
( 24) FMDEMON          Minor Unexpected Event   dbgPort
( 25) NTP              Minor Unexpected Event   dbgPort
( 26) WEDDEMON         Minor Unexpected Event   dbgPort
( 27) SNMPTRAP         Minor Unexpected Event   dbgPort
( 28) DLMM             Minor Unexpected Event   dbgPort
( 29) TR069DEMON       Minor Unexpected Event   dbgPort
( 30) PSTN             Minor Unexpected Event   dbgPort
```

Figure 3-106 Spy command

Table 3-68 Spy level

Spy Level	Description
0	GEN Level: superfluous information
1	Function Enter Level: entering function
2	Event Level: normal and expected event(i.e. link up)
3(Default)	Minor Error Level: minor unexpected event
4	Major Error Level: major unexpected event
5	Fatal Error Level: fatal error
6	Off Level: Level to turn off Spy Information

Chapter 4. Edit Configuration File

This Chapter describes the format of VENUS 29xx Series configuration file. The configuration file is actually composed of VENUS 29xx Series configuration parameters. Users are allowed to edit this file for their own need. By downloading the configuration file from a TFTP server to the device, user can easily perform the system configuration and don't have to go through the step-by-step configuration items provided by the terminal interface. User can upload the configuration file to TFTP server through menu tree. Please refer to Section 3.5.3.2 for information on configuration upload and download.

4.1 File Format

The configuration file must be in ASCII text format, use the text mode editor like Microsoft Notepad, or Microsoft WordPad to edit the file. The rules of the file are:

- The file must start with the string "IAD Configuration File" at the 1st line (case sensitive).
- The parameters must be organized by its category name. The category name is defined as: "[" + menu name + "_" + sub-menu name + "]". Such as "[interface_pstn]".
- The parameter format defined as: item name + "=" + value. Such as "registrar_addr_0=192.168.1.217".

4.2 Example File

The following is an example configuration file.

IAD Configuration File

[interface_ethernet]

ethernet_mode_0=1

ip_mode_0=0

ip_address_0=192.168.1.222

net_mask_0=255.255.255.0

default_gateway_0=192.168.1.20

dns1_address_0=0.0.0.0

dns2_address_0=0.0.0.0

dhcp_iad_0=0

dhcp_dns_0=0

dhcp_server_addr1_0=0.0.0.0

dhcp_net_mask1_0=0.0.0.0

dhcp_server_addr2_0=0.0.0.0

dhcp_net_mask2_0=0.0.0.0

pppoe_user_name_0=user

pppoe_user_pwd_0=123

pppoe_dns_0=0

pppoe_gateway_0=1

pppoe_reconnect_0=1

mtu_size_0=1500

ethernet_mode_1=0

ip_mode_1=0

ip_address_1=0.0.0.0

net_mask_1=0.0.0.0

default_gateway_1=0.0.0.0

dns1_address_1=0.0.0.0

dns2_address_1=0.0.0.0

dhcp_iad_1=0

dhcp_dns_1=0

dhcp_server_addr1_1=0.0.0.0

dhcp_net_mask1_1=0.0.0.0

dhcp_server_addr2_1=0.0.0.0

dhcp_net_mask2_1=0.0.0.0

pppoe_user_name_1=

pppoe_user_pwd_1=

pppoe_dns_1=0

pppoe_gateway_1=1

pppoe_reconnect_1=1

mtu_size_1=1500

[interface_pstn]

seizure_prefix=#*1

ring_off_time=40

lifeline_channel=1

ring_counter=0

seizure_prefix=#*1

ring_off_time=40

lifeline_channel=1

ring_counter=0

```
[interface_analog]
    user_number_0=1000
    user_name_0=
    register_mode_0=2
    min_flash_time_0=50
    max_flash_time_0=500
    line_reversal_0=1
    caller_id_mode_0=0
    hotline_number_0=
    hotline_timeout_0=0
    realm_index_0=0
    local_port_0=5060
    user_auth_id_0=
    user_auth_pwd_0=
    digitmap_0=x.
    always_number_0=
    busy_number_0=
    no_answer_number_0=
    wait_times_0=3
    call_id_block_0=0
    prefer_call_id_0=
    rx_gain_0=0
    tx_gain_0=0
    proto_0=6
    last_set_state_0=2
```

pref_fax_coding_0=1

companding_0=4

ec_config_0=1

fxs_offhook_db_0=20

fxs_onhook_db_0=20

fxs_seize_detect_0=150

fxs_orig_clear_detect_0=300

fxs_answ_clear_detect_0=300

fxs_cpc_wait_0=200

fxs_cpc_dur_0=850

fxs_ring_id_0=255

fxs_cid_type_0=1

fxs_local_batt_conn_rev_0=1

fxs_rem_batt_conn_rev_0=1

fxs_batt_ring_rev_0=0

answer_wait_0=65535

call_limit_0=65535

fax_holdover_0=2000

hangup_wait_0=65535

out_wait_0=400

out_type_0=0

tone_out_on_time_0=200

tone_out_off_time_0=200

tone_out_power_0=-130

pause_duration_0=1000

plar_addr_len_0=0

plar_addr_digits_0=

user_number_1=1001

user_name_1=

register_mode_1=2

min_flash_time_1=50

max_flash_time_1=500

line_reversal_1=1

caller_id_mode_1=0

hotline_number_1=

hotline_timeout_1=0

realm_index_1=0

local_port_1=5060

user_auth_id_1=

user_auth_pwd_1=

digitmap_1=x.

always_number_1=

busy_number_1=

no_answer_number_1=

wait_times_1=3

call_id_block_1=0

prefer_call_id_1=

rx_gain_1=0

tx_gain_1=0

proto_1=6

last_set_state_1=2

pref_fax_coding_1=1

companding_1=4

ec_config_1=1

fxs_offhook_db_1=20

fxs_onhook_db_1=20

fxs_seize_detect_1=150

fxs_orig_clear_detect_1=300

fxs_answ_clear_detect_1=300

fxs_cpc_wait_1=200

fxs_cpc_dur_1=850

fxs_ring_id_1=255

fxs_cid_type_1=1

fxs_local_batt_conn_rev_1=1

fxs_rem_batt_conn_rev_1=1

fxs_batt_ring_rev_1=0

answer_wait_1=65535

call_limit_1=65535

fax_holdover_1=2000

hangup_wait_1=65535

out_wait_1=400

out_type_1=0

tone_out_on_time_1=200

tone_out_off_time_1=200

tone_out_power_1=-130

pause_duration_1=1000

plar_addr_len_1=0

plar_addr_digits_1=

user_number_2=1002

user_name_2=

register_mode_2=2

min_flash_time_2=50

max_flash_time_2=500

line_reversal_2=1

caller_id_mode_2=0

hotline_number_2=

hotline_timeout_2=0

realm_index_2=0

local_port_2=5060

user_auth_id_2=

user_auth_pwd_2=

digitmap_2=x.

always_number_2=

busy_number_2=

no_answer_number_2=

wait_times_2=3

call_id_block_2=0

prefer_call_id_2=

rx_gain_2=0

tx_gain_2=0

proto_2=6

last_set_state_2=2

pref_fax_coding_2=1

companding_2=4

ec_config_2=1

fxs_offhook_db_2=20

fxs_onhook_db_2=20

fxs_seize_detect_2=150

fxs_orig_clear_detect_2=300

fxs_answ_clear_detect_2=300

fxs_cpc_wait_2=200

fxs_cpc_dur_2=850

fxs_ring_id_2=255

fxs_cid_type_2=1

fxs_local_batt_conn_rev_2=1

fxs_rem_batt_conn_rev_2=1

fxs_batt_ring_rev_2=0

answer_wait_2=65535

call_limit_2=65535

fax_holdover_2=2000

hangup_wait_2=65535

out_wait_2=400

out_type_2=0

tone_out_on_time_2=200

tone_out_off_time_2=200

tone_out_power_2=-130

pause_duration_2=1000

plar_addr_len_2=0

plar_addr_digits_2=

user_number_3=1003

user_name_3=

register_mode_3=2

min_flash_time_3=50

max_flash_time_3=500

line_reversal_3=1

caller_id_mode_3=0

hotline_number_3=

hotline_timeout_3=0

realm_index_3=0

local_port_3=5060

user_auth_id_3=

user_auth_pwd_3=

digitmap_3=x.

always_number_3=

busy_number_3=

no_answer_number_3=

wait_times_3=3

call_id_block_3=0

prefer_call_id_3=

rx_gain_3=0

tx_gain_3=0

proto_3=6

last_set_state_3=2

pref_fax_coding_3=1

companding_3=4

ec_config_3=1

fxs_offhook_db_3=20

fxs_onhook_db_3=20

fxs_seize_detect_3=150

fxs_orig_clear_detect_3=300

fxs_answ_clear_detect_3=300

fxs_cpc_wait_3=200

fxs_cpc_dur_3=850

fxs_ring_id_3=255

fxs_cid_type_3=1

fxs_local_batt_conn_rev_3=1

fxs_rem_batt_conn_rev_3=1

fxs_batt_ring_rev_3=0

answer_wait_3=65535

call_limit_3=65535

fax_holdover_3=2000

hangup_wait_3=65535

out_wait_3=400

out_type_3=0

tone_out_on_time_3=200

tone_out_off_time_3=200

tone_out_power_3=-130

pause_duration_3=1000

plar_addr_len_3=0

plar_addr_digits_3=

user_number_4=1004

user_name_4=

register_mode_4=2

min_flash_time_4=50

max_flash_time_4=500

line_reversal_4=1

caller_id_mode_4=0

hotline_number_4=

hotline_timeout_4=0

realm_index_4=0

local_port_4=5060

user_auth_id_4=

user_auth_pwd_4=

digitmap_4=x.

always_number_4=

busy_number_4=

no_answer_number_4=

wait_times_4=3

call_id_block_4=0

prefer_call_id_4=

rx_gain_4=0

tx_gain_4=0

proto_4=6

last_set_state_4=2

pref_fax_coding_4=1

companding_4=4

ec_config_4=1

fxs_offhook_db_4=20

fxs_onhook_db_4=20

fxs_seize_detect_4=150

fxs_orig_clear_detect_4=300

fxs_answ_clear_detect_4=300

fxs_cpc_wait_4=200

fxs_cpc_dur_4=850

fxs_ring_id_4=255

fxs_cid_type_4=1

fxs_local_batt_conn_rev_4=1

fxs_rem_batt_conn_rev_4=1

fxs_batt_ring_rev_4=0

answer_wait_4=65535

call_limit_4=65535

fax_holdover_4=2000

hangup_wait_4=65535

out_wait_4=400

out_type_4=0

tone_out_on_time_4=200

tone_out_off_time_4=200

tone_out_power_4=-130

pause_duration_4=1000

plar_addr_len_4=0

plar_addr_digits_4=

user_number_5=1005

user_name_5=

register_mode_5=2

min_flash_time_5=50

max_flash_time_5=500

line_reversal_5=1

caller_id_mode_5=0

hotline_number_5=

hotline_timeout_5=0

realm_index_5=0

local_port_5=5060

user_auth_id_5=

user_auth_pwd_5=

digitmap_5=x.

always_number_5=

busy_number_5=

no_answer_number_5=

wait_times_5=3

call_id_block_5=0

prefer_call_id_5=

rx_gain_5=0

tx_gain_5=0

proto_5=6

last_set_state_5=2

pref_fax_coding_5=1

companding_5=4

ec_config_5=1

fxs_offhook_db_5=20

fxs_onhook_db_5=20

fxs_seize_detect_5=150

fxs_orig_clear_detect_5=300

fxs_answ_clear_detect_5=300

fxs_cpc_wait_5=200

fxs_cpc_dur_5=850

fxs_ring_id_5=255

fxs_cid_type_5=1

fxs_local_batt_conn_rev_5=1

fxs_rem_batt_conn_rev_5=1

fxs_batt_ring_rev_5=0

answer_wait_5=65535

call_limit_5=65535

fax_holdover_5=2000

hangup_wait_5=65535

out_wait_5=400

out_type_5=0

tone_out_on_time_5=200

tone_out_off_time_5=200

tone_out_power_5=-130

pause_duration_5=1000

plar_addr_len_5=0

plar_addr_digits_5=

user_number_6=1006

user_name_6=

register_mode_6=2

min_flash_time_6=50

max_flash_time_6=500

line_reversal_6=1

caller_id_mode_6=0

hotline_number_6=

hotline_timeout_6=0

realm_index_6=0

local_port_6=5060

user_auth_id_6=

user_auth_pwd_6=

digitmap_6=x.

always_number_6=

busy_number_6=

no_answer_number_6=

wait_times_6=3

call_id_block_6=0

prefer_call_id_6=

rx_gain_6=0

tx_gain_6=0

proto_6=6

last_set_state_6=2

pref_fax_coding_6=1

companding_6=4

ec_config_6=1

fxs_offhook_db_6=20

fxs_onhook_db_6=20

fxs_seize_detect_6=150

fxs_orig_clear_detect_6=300

fxs_answ_clear_detect_6=300

fxs_cpc_wait_6=200

fxs_cpc_dur_6=850

fxs_ring_id_6=255

fxs_cid_type_6=1

fxs_local_batt_conn_rev_6=1

fxs_rem_batt_conn_rev_6=1

fxs_batt_ring_rev_6=0

answer_wait_6=65535

call_limit_6=65535

fax_holdover_6=2000

hangup_wait_6=65535

out_wait_6=400

out_type_6=0

tone_out_on_time_6=200

tone_out_off_time_6=200

tone_out_power_6=-130

pause_duration_6=1000

plar_addr_len_6=0

plar_addr_digits_6=

user_number_7=1007

user_name_7=

register_mode_7=2

min_flash_time_7=50

max_flash_time_7=500

line_reversal_7=1

caller_id_mode_7=0

hotline_number_7=

hotline_timeout_7=0

realm_index_7=0

local_port_7=5060

user_auth_id_7=

user_auth_pwd_7=

digitmap_7=x.

always_number_7=

busy_number_7=

no_answer_number_7=

wait_times_7=3

call_id_block_7=0

prefer_call_id_7=

rx_gain_7=0

tx_gain_7=0

proto_7=6

last_set_state_7=2

pref_fax_coding_7=1

companding_7=4

ec_config_7=1

fxs_offhook_db_7=20

fxs_onhook_db_7=20

fxs_seize_detect_7=150

fxs_orig_clear_detect_7=300

fxs_answ_clear_detect_7=300

fxs_cpc_wait_7=200

fxs_cpc_dur_7=850

fxs_ring_id_7=255

fxs_cid_type_7=1

fxs_local_batt_conn_rev_7=1

fxs_rem_batt_conn_rev_7=1

fxs_batt_ring_rev_7=0

answer_wait_7=65535

call_limit_7=65535

fax_holdover_7=2000

hangup_wait_7=65535

out_wait_7=400

out_type_7=0

tone_out_on_time_7=200

tone_out_off_time_7=200

tone_out_power_7=-130

pause_duration_7=1000

plar_addr_len_7=0

plar_addr_digits_7=

[system_general]

device_name=00:90:bb:00:03:7c

serial_number=

sys_server_addr=

clone_mac_addr=

signal_tos=7

restart_wait=30

op_mode=13

```
[system_ntp]
    ntp_mode=1
    ntp_server_addr=192.168.1.218
    ntp_server_port=123
    polling_interval=1024
    time_zone=-8

[system_usermgt]
    admin_name=admin
    admin_pswd=21232f297a57a5a743894a0e4a801fc3
    operator_name=user
    operator_pswd=d41d8cd98f00b204e9800998ecf8427e
    monitor_name=guest
    monitor_pswd=d41d8cd98f00b204e9800998ecf8427e
    use_radius_login=0
    radius_server_addr=
    radius_server_port=0
    radius_share_secret=

[system_ui]
    telnet_server_mode=0
    telnet_server_port=23
    telnet_idle_timeout=64
    telnet_interface=0
```

web_server_mode=1

web_server_port=80

web_idle_timeout=20

web_interface=1

[system_provision]

prov_server_addr=

prov_server_port=69

prov_protocal=0

prov_user_name=

prov_user_pswd=

prov_file_name=

file_encryption=0

prov_timeout=0

[system_snmp]

snmp_server_mode=0

snmp_server_port=161

snmp_trap_mode=0

snmp_trap_port=162

snmp_cdrtrap_mode=0

system_name=

system_location=

system_contact=

get_community=

```
        set_community=
        trap_community=
        nat_pass_mode=0
        nat_pass_interval=60
        trap_dest_addr1=
        trap_dest_addr2=
        trap_dest_addr3=
        trap_dest_addr4=
    [system_tr069]
        acsurl=
      cpeport=
    mode=
    [sip_general]
        fqdn_mode=0
        fqdn_name=
        dns_query_type=0
        trust_proxy=0
        t1_time=500
        regst_auth=0
        pwd_with_md5=0
        use_port_url=0
        compact_mode=0
        invite_no_sdp=0
        band_ring_bktone=0
        invite_timeout=60
```

session_expire_mode=0

session_expire_time=3600

min_session_expire=120

prack_mode=0

info_mode=0

subscrip_mode=0

pound_symbol_mode=0

ring_mode=0

discon_on_broken=0

broken_conn_timeout=60

current_disconnect=0

fallback_rout_table=0

prefer_rout_table=0

critical_digit_timer=4

partial_digit_timer=16

[sip_realm]

realm_valid_0=1

domain_name_0=

proxy_addr_0=192.168.1.217

proxy_port_0=5060

redu_proxy_addr_0=

redu_proxy_port_0=0

registrar_addr_0=192.168.1.217

registrar_port_0=5060

backup_reg_addr_0=

backup_reg_port_0=0

registration_time_0=3600

reregist_timing_0=100

outbd_proxy_addr_0=

outbd_proxy_port_0=5060

transport_way_0=0

aor_name_0=

keep_alive_mode_0=0

keep_alive_interval_0=60

realm_valid_1=0

domain_name_1=

proxy_addr_1=

proxy_port_1=5060

redu_proxy_addr_1=

redu_proxy_port_1=0

registrar_addr_1=

registrar_port_1=5060

backup_reg_addr_1=

backup_reg_port_1=0

registration_time_1=3600

reregist_timing_1=100

outbd_proxy_addr_1=

outbd_proxy_port_1=5060

transport_way_1=0

aor_name_1=

keep_alive_mode_1=0

keep_alive_interval_1=60

realm_valid_2=0

domain_name_2=

proxy_addr_2=

proxy_port_2=5060

redu_proxy_addr_2=

redu_proxy_port_2=0

registrar_addr_2=

registrar_port_2=5060

backup_reg_addr_2=

backup_reg_port_2=0

registration_time_2=3600

reregist_timing_2=100

outbd_proxy_addr_2=

outbd_proxy_port_2=5060

transport_way_2=0

aor_name_2=

keep_alive_mode_2=0

keep_alive_interval_2=60

realm_valid_3=0

domain_name_3=

proxy_addr_3=

proxy_port_3=5060

redu_proxy_addr_3=

redu_proxy_port_3=0

registrar_addr_3=

registrar_port_3=5060

backup_reg_addr_3=

backup_reg_port_3=0

registration_time_3=3600

reregist_timing_3=100

outbd_proxy_addr_3=

outbd_proxy_port_3=5060

transport_way_3=0

aor_name_3=

keep_alive_mode_3=0

keep_alive_interval_3=60

realm_valid_4=0

domain_name_4=

proxy_addr_4=

proxy_port_4=5060

redu_proxy_addr_4=

redu_proxy_port_4=0

registrar_addr_4=

registrar_port_4=5060

backup_reg_addr_4=

backup_reg_port_4=0

registration_time_4=3600

reregist_timing_4=100

outbd_proxy_addr_4=

outbd_proxy_port_4=5060

transport_way_4=0

aor_name_4=

keep_alive_mode_4=0

keep_alive_interval_4=60

realm_valid_5=0

domain_name_5=

proxy_addr_5=

proxy_port_5=5060

redu_proxy_addr_5=

redu_proxy_port_5=0

registrar_addr_5=

registrar_port_5=5060

backup_reg_addr_5=

backup_reg_port_5=0

registration_time_5=3600

reregist_timing_5=100

outbd_proxy_addr_5=

outbd_proxy_port_5=5060

transport_way_5=0

aor_name_5=

keep_alive_mode_5=0

keep_alive_interval_5=60

realm_valid_6=0

domain_name_6=

proxy_addr_6=

proxy_port_6=5060

redu_proxy_addr_6=

redu_proxy_port_6=0

registrar_addr_6=

registrar_port_6=5060

backup_reg_addr_6=

backup_reg_port_6=0

registration_time_6=3600

reregist_timing_6=100

outbd_proxy_addr_6=

outbd_proxy_port_6=5060

transport_way_6=0

aor_name_6=

keep_alive_mode_6=0

keep_alive_interval_6=60

realm_valid_7=0

domain_name_7=

proxy_addr_7=

proxy_port_7=5060

redu_proxy_addr_7=

redu_proxy_port_7=0

registrar_addr_7=

registrar_port_7=5060

backup_reg_addr_7=

backup_reg_port_7=0

registration_time_7=3600

reregist_timing_7=100

outbd_proxy_addr_7=

outbd_proxy_port_7=5060

transport_way_7=0

aor_name_7=

keep_alive_mode_7=0

keep_alive_interval_7=60

[sip_group]

group_number_0=10

group_name_0=

register_mode_0=0

realm_index_0=0

local_port_0=5060

group_auth_id_0=

group_auth_pwd_0=

involved_tcids_0=0

group_number_1=11

group_name_1=

register_mode_1=0

realm_index_1=0

local_port_1=5060

group_auth_id_1=

group_auth_pwd_1=

involved_tcids_1=0

group_number_2=12

group_name_2=

register_mode_2=0

realm_index_2=0

local_port_2=5060

group_auth_id_2=

group_auth_pwd_2=

involved_tcids_2=0

group_number_3=13

group_name_3=

register_mode_3=0

realm_index_3=0

local_port_3=5060

group_auth_id_3=

group_auth_pwd_3=

involved_tcids_3=0

group_number_4=14

group_name_4=

register_mode_4=0

realm_index_4=0

local_port_4=5060

group_auth_id_4=

group_auth_pwd_4=

involved_tcids_4=0

group_number_5=15

group_name_5=

register_mode_5=0

realm_index_5=0

local_port_5=5060

group_auth_id_5=

group_auth_pwd_5=

involved_tcids_5=0

group_number_6=16

group_name_6=

register_mode_6=0

realm_index_6=0

local_port_6=5060

group_auth_id_6=

group_auth_pwd_6=

involved_tcids_6=0

group_number_7=17

group_name_7=

register_mode_7=0

realm_index_7=0

local_port_7=5060

group_auth_id_7=

group_auth_pwd_7=

involved_tcids_7=0

[sip_phonerout]

dest_phone_prefix_0=

src_phone_prefix_0=

sip_ua_addr_0=

sip_ua_port_0=0

[advanced_rtp]

rtp_base_port=30000

rtcp_mode=0

rtp_filter_mode=0

srtp_mode=0

srtp_crypto_suites=0

audio_codec=0

g711_ptime=20

g723_ptime=30

g726_ptime=20

g729_ptime=30

use_rfc2833=0

rfc2833_ptype=101

use_g726_codec=0

g726_16_ptype=2

g726_24_ptype=2

g726_32_ptype=2

g726_40_ptype=2

vad_mode=0

echo_cancel_mode=0

rtcp_window_size=32

rtcp_interval=15

service_class=0

security_mode=0

uncryption=0

[advanced_call]

clip_mode=0

dtmf_prefix=D

digit_timeout=4

call_wait_interval=3

call_wait_times=3

deny_2nd_dailing=0

max_call=0

e164_mode=0

dialing_prefix=

emerg_call_num1=

emerg_call_num2=

emerg_call_num3=

emerg_call_num4=

emerg_call_mode=0

call_hold=0

call_hold_mode=0

call_waiting=0

three_way_mode=0

confer_server_url=

attended_transfer=0

call_wait_deactive=*70

call_id_active=*67

af_active_prefix=*72

af_deactive_prefix=*73

call_return_function=*69

call_redail_function=*68

unattended_transfer=*66

call_park_function=*61

call_pickup_function=*62

call_park_server=

call_intercept=*63

fax_signal_method=1

t30_fax_codec=0

fax_tone_detection=0

modem_method=1

modem_codec=0

modem_tone_detect=0

[advanced_region]

region_id=2

region_code=

area_code=

international_code=

domestic_code=

ring_frequency=25

ring_cadence_0=10,40,0,0,0,0,180000

ring_cadence_1=20,40,0,0,0,0,180000

ring_cadence_2=8,4,8,40,0,0,180000

ring_cadence_3=4,2,4,2,8,40,180000

ring_cadence_4=3,2,10,2,3,40,180000

ring_cadence_5=5,1,0,0,0,0,600

ring_cadence_6=5,2,3,2,5,30,180000

ring_cadence_7=20,40,0,0,0,0,180000

ring_cadence_8=30,50,0,0,0,0,180000

ring_cadence_9=5,1,0,0,0,0,600

ring_cadence_10=10,40,0,0,0,0,180000

tone_timeout_0=40000

tone_numrow_0=2

tone_0_0=1,450,-100,0,0,0,0,0,0,350

tone_0_1=0,0,0,0,0,0,0,0,0,350

tone_timeout_1=180000

tone_numrow_1=2

tone_1_0=1,450,-100,0,0,0,0,0,0,1000

tone_1_1=0,0,0,0,0,0,0,0,0,4000

tone_timeout_2=40000

tone_numrow_2=2

tone_2_0=1,450,-100,0,0,0,0,0,0,700

tone_2_1=0,0,0,0,0,0,0,0,0,700

tone_timeout_3=16000

tone_numrow_3=1

tone_3_0=1,450,-100,0,0,0,0,0,0,-1

tone_timeout_4=16000

tone_numrow_4=1

tone_4_0=1,400,-100,0,0,0,0,0,0,-1

tone_timeout_5=-1

```
tone_numrow_5=4
tone_5_0=1,450,-100,0,0,0,0,0,0,100
tone_5_1=0,0,0,0,0,0,0,0,0,100
tone_5_2=1,450,-100,0,0,0,0,0,0,400
tone_5_3=0,0,0,0,0,0,0,0,0,400
tone_timeout_6=0
tone_numrow_6=4
tone_6_0=1,600,-240,0,0,0,0,0,0,125
tone_6_1=0,0,0,0,0,0,0,0,0,125
tone_6_2=1,600,-240,0,0,0,0,0,0,125
tone_6_3=0,0,0,0,0,0,0,0,0,-1
tone_timeout_7=-1
tone_numrow_7=4
tone_7_0=1,450,-200,0,0,0,0,0,0,250
tone_7_1=0,0,0,0,0,0,0,0,0,250
tone_7_2=1,450,-200,0,0,0,0,0,0,250
tone_7_3=0,0,0,0,0,0,0,0,0,-1
tone_timeout_8=60000
tone_numrow_8=1
tone_8_0=1,950,0,0,0,0,0,0,0,-1
tone_timeout_9=-1
tone_numrow_9=1
tone_9_0=1,950,0,0,0,0,0,0,0,-1
tone_timeout_10=16000
tone_numrow_10=1
```

tone_10_0=1,450,-100,0,0,0,0,0,0,-1

tone_timeout_11=600

tone_numrow_11=2

tone_11_0=1,450,-100,0,0,0,0,0,0,400

tone_11_1=0,0,0,0,0,0,0,0,0,40

tone_timeout_12=635

tone_numrow_12=4

tone_12_0=1,450,-200,0,0,0,0,0,0,400

tone_12_1=0,0,0,0,0,0,0,0,0,100

tone_12_2=2,2130,-150,2750,-150,0,0,0,0,85

tone_12_3=0,0,0,0,0,0,0,0,0,5000

tone_timeout_13=30000

tone_numrow_13=2

tone_13_0=1,450,-100,0,0,0,0,0,0,350

tone_13_1=0,0,0,0,0,0,0,0,0,350

tone_timeout_14=0

tone_numrow_14=1

tone_14_0=1,250,0,0,0,0,0,0,0,50

tone_timeout_15=0

tone_numrow_15=1

tone_15_0=1,350,0,0,0,0,0,0,0,5000

tone_timeout_16=-1

tone_numrow_16=2

tone_16_0=1,450,-100,0,0,0,0,0,0,700

tone_16_1=0,0,0,0,0,0,0,0,0,700

```
        tone_timeout_17=-1
        tone_numrow_17=4
        tone_17_0=1,400,-130,0,0,0,0,0,0,250
        tone_17_1=0,0,0,0,0,0,0,0,0,250
        tone_17_2=1,400,-130,0,0,0,0,0,0,250
        tone_17_3=0,0,0,0,0,0,0,0,0,3250

    [advanced_nat]
        nat_ip_addr=0.0.0.0
        stun_server_addr=
        stun_server_port=0
        polling_interval=64

    [advanced_ddns]
        ddns_mode=0
        ddns_server_addr=
        host_name=
        user_id=
        user_pswd=

    [advanced_dsp]
        allocation_method=0
        long_loop_mode=0

    [data_routing]
```

```
        dest_ip_addr_0=0.0.0.0
        dest_net_mask_0=0.0.0.0
        gateway_addr_0=0.0.0.0
        metric_0=0

    [data_natmap]
        nat_mode=1
        protocol_mode_0=0
        internal_ip_addr_0=0.0.0.0
        inter_serv_port_0=0
        exter_serv_port_0=0

    [data_dhcp]
        dhcp_mode=0
        dhcp_net_addr=0.0.0.0
        dhcp_net_mask=0.0.0.0
        domain_name=
        dns_ip_addr0=0.0.0.0
        dns_ip_addr1=0.0.0.0
        dns_ip_addr2=0.0.0.0
        dns_ip_addr3=0.0.0.0
        gw_ip_addr0=0.0.0.0
        gw_ip_addr1=0.0.0.0
        gw_ip_addr2=0.0.0.0
        gw_ip_addr3=0.0.0.0
```

default_lease_time=0

free_ip_low1=0.0.0.0

free_ip_low2=0.0.0.0

free_ip_low3=0.0.0.0

free_ip_low4=0.0.0.0

free_ip_low5=0.0.0.0

free_ip_high1=0.0.0.0

free_ip_high2=0.0.0.0

free_ip_high3=0.0.0.0

free_ip_high4=0.0.0.0

free_ip_high5=0.0.0.0

assign_host_name_0=

assign_ip_addr_0=0.0.0.0

assign_mac_addr_0=

[data_qos]

qos_mode=0

signal_dscp=37

voice_dscp=37

mgt_dscp=0

data_dscp=0

signal_tos=6

voice_tos=6

mgt_tos=0

data_tos=0

```
[data_vlan]
    vlan_mode=0
    signal_vlan_id=0
    voice_vlan_id=0
    mgt_vlan_id=0
    data_vlan_id=0
    signal_vlan_prior=0
    voice_vlan_prior=0
    mgt_vlan_prior=0
    data_vlan_prior=0
    vlan_tag_0=0
    vlan_tag_1=0
    vlan_vid_0=0
    vlan_vid_1=0
    vlan_vid_2=0
    vlan_priority_0=0
    vlan_priority_1=0
    vlan_priority_2=0
    vlan_group_0=7
    vlan_group_1=0
    vlan_group_2=0

[fm_threshold]
    cpu_threshold=100
```

```
ram_threshold=100

dsp_threshold=100

# end of configuration file
```

Chapter 5. Troubleshooting

5.1 Instruments

The following instruments may help to allocate the problem:

- **A multi-meter**: to identify the line condition, the power condition, etc.
- **A PC**: with a LAN card installed and the IP setup configured properly
- **A network scope**: to identify the network status and the traffic load over the network.

5.2 Troubleshooting

Turn off the power first. Follow the procedures below to allocate the problem:

1) Cable Connections

Make sure all the cables are connected correctly and firmly. Check if you provide a correct power source.

2) The Power Source

The LED indicators offer some helpful information for users to check the hardware status of VENUS 29xx Series. Check if the PWR LED is ON. Use the multi-meter to measure the power supply.

3) Boot up Diagnosis

Observe the boot up sequence of the gateway. During the boot up phase, VENUS 29xx Series performs the hardware initialization, run-time AP software verification, interface module detection and then the hardware diagnosis procedure. Check if all the LEDs can display correctly.

The SYS ALM LED indicates the diagnosis state: while it is Yellow, the device is performing diagnosis. A Red SYS ALM LED indicates a hardware failure. A Green SYS LED indicates no hardware failure.

Login to the Craft port Interface, and select the “Hardware Configuration and Diagnosis Report” menu to see if the device reports correct hardware configuration, type of interface cards installed, number of voice channel detected, and the diagnosis results.

In case of hardware failures, record the indication of LEDs during the boot up phase, and report the diagnosis result generated by VENUS 29xx Series to our customer service.

4) IP Network Interface Connection

Check the indicator for the WAN interface indicating normal condition. If the 10/100Base-T Fast Ethernet port is used, check if the 10/100BT LED is on. Verify if a correct Ethernet cable is used. To connect VENUS 29xx Series directly to another Ethernet port of a network node (for example a router), a cross cable, instead of a straight cable, may be used.

5) IP Address Configuration

Check if the IP address was assigned statically or obtained dynamically from a DHCP server.

If the DHCP client mode is enabled, look at the DHCP server for the exact IP address assigned to the gateway or monitor the display message of the DHCP server discovery status from the Terminal User Interface via the Craft port. Remember that the DHCP server must be located in the local network where the DHCP server can receive the broadcast packets sent from the gateway. Make sure there is unused IP address available on the DHCP server.

If static IP address is enabled, make sure the net mask and the default Gateway are set correctly. Use the PC to perform a PING test to the gateway, or activate the PING feature from within the Craft port Interface to verify the IP packet transmission between the gateway and some other nodes. Try to PING the default gateway first, then the SIP Proxy Server. Check with your access service provider for a well-maintained WAN link.

Use the Network Scope (or the Protocol Analyzer) to monitor the packets sent received by the VENUS 29xx Series.

6) Firewall and NAT

Normally, a public IP address should be used for VENUS 29xx Series. In case the VENUS 29xx Series is located in the local network behind the firewall or router, Make sure the IP packets is not blocked or modified.

For the security purpose, a firewall server or VPN is usually installed to filter out unauthorized accesses from the outside world. Make sure the traffic to/from the VoIP gateway is not blocked.

The NAT server may cause problem as well. The NAT server performs the Network Address Translation between the public IP address and the private IP address. It may not recognize some Text-based protocols (like SIP) used by the VoIP gateway and may cause incorrect IP address or port translation for a packet.

7) Off-hook the Telephone Handset

Pick-up the phone and hear to the response. Check if the LED of the telephone port is turned ON. If no tone is heard, check the transmission between the Proxy and Registration Server and the gateway again.

8) Make a Local Call

If the dial tone can be heard, try to dial a local number to another port on the same gateway. See it the ring back tone is heard and the called party are ringing. See if the Channel LED flashes while the phone rings. Answer the phone, and start conversation. Hear the voice quality.

9) Make an On-Net Call

Try to make a call to another VoIP gateway somewhere located in the IP network. If the ring back tone is heard, but no voice can be heard after the called party answers the phone, the voice packets sent by these two gateways may be blocked somehow (reason like firewall, VPN, etc). Contact your service provider to check the setup of the routing path.

The voice waves are converted to digital voice coding (such as G.711, G.723, G.729, etc) and encapsulated in RTP packets. RTP is a transport protocol running on the top of UDP. To make a VoIP phone call, a RTP connection is created for the caller and the called gateway. The UDP port used for RTP is not fixed. VENUS 29xx Series uses the UDP number started from 30000.

10) Hear the Voice Quality

If the voice quality is not good, possible reasons include

Low WAN link throughput not able to support enough bandwidth for voice traffic

(see the following table for calculation of maximum bandwidth required)

Network congestion: a well maintained network flow control policy or protocol help to control the traffic and prioritize the service for different type of media

Type of voice codec

Network device delay: firewall, router, switch, access device, … etc all contribute latency to the traffic. A good network configuration in advance is very important.

Type of the Voice Codec selected is a big factor and tradeoff to the voice quality. The PCM type codec (G.711 u-law or A-law) provide excellent voice quality, but consume large amount of network bandwidth. A higher compression rate codec provide fair voice quality and consumes less network bandwidth, however requires more CPU/DSP power which increase the voice latency.

Table 5-1 Voice Codec Bandwidth

Codec Types	**Bit Rate**
ITU-T G.711 A-Law PCM	64K
ITU-T G.711 μ-Law PCM	64K
ITU-T G.723.1	6.3K/5.3K
ITU-T G.729A	8K
ITU-T G.726.	16/24/32/40K

Appendix A Supplemental Telephone Features

For the traditional PSTN service providers, they used to invest on extra equipment or technology for those services and may charge extra to their customers.

The supplemental telephone features like Call Forwarding, Call Waiting, Caller ID Display, Caller ID Blocking, Speed Dial, Call Return and 3-way Conference can be provided.

The following Table describes some example procedures for VENUS 29xx Series to enable or disable several example supplemental features (The following example needs SIP Proxy server support).

Table A-1 Supplemental Telephony Features

Features	Description	Examples
Call Forward All (CFA)	Forward all the incoming calls to a designated telephone number	All installed users can assign an always call forward number by menu tree or CLI. Once a new call incoming, the conversation will be forwarded to the specified user through the response 302.
Call Forward Busy (CFB)	Forward all the incoming calls while the local phone is busy	All installed users can assign a busy call forward number by menu tree or CLI. Once a new call incoming and under busy, the conversation will be forwarded to the specified user through response 302.
Call Forward No Answer (CFNA)	Forward the incoming calls to a preset telephone number if the call is not answered after several rings	All installed users can assign a No answer call forward number and ringing timeout time by menu tree or CLI. Once a new call incoming and ringing timeout, the conversation will be forwarded to the specified user through response 302.
Caller ID Blocking	The Caller ID (telephone number) and the registered Name of the local phone will not be displayed on the Called phone for privacy.	The caller can hide its caller ID by selection this feature.
Caller ID Display	The Caller ID (telephone number) and the registered Name will be displayed on	The called user can display caller ID by different mode in different country.

Features	Description	Examples
	the Called phone bye three mode: FSK, ETSI and DTMF. User also can disable the caller ID display function by selection mode Disable.	
Call Return	To call back to the caller of the last incoming call. This feature works even if last incoming call was not answered.	To make a return call, please pickup the phone and press call return digits and default value is *69.
Call Waiting	This feature allows interruption of a 3rd party to a call in progress. Make the incoming call to wait until current conversation finished. The talking person will be informed by short tone	User can hear call waiting tone when a new incoming call under a conversation. He can press flash hook to hold the current conversation partner. Pressing another flash hook will hold the 3rd party.
3-Way conference	This feature allows user to make two call calls and mixed the RTP stream and make a 3 party joined conference.	User A makes a conversation with user B. User A presses flash hook to hold user B. User A can dial user C to build second call. User A presses flash hook to start three-way conference. User A can press flash hook to disable User C conversation. Three-way conference can’t be built in User C ringing status for the current version.
Call transfer	This feature allow user to make a consultation or un-consultation call transfer.	User A makes a conversation with User B. User A press flash hook to hold User B. User A dial User C to make the second conversation. User A hang up the phone set when ringing for un-consultation call transfer or when call built for consultation call transfer. User B and User C make another call after User A hangs up.

Appendix B Abbreviation

Abbreviation	Stands for
ARP	Address Resolution Protocol
BootP	Bootstrap Protocol
CLI	Command Line Interface
CPU	Central Processing Unit
DDN	Digital Data Network
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System or Domain Name Server
DSP	Digital Signal Processor
FXS	Foreign Exchange Station
HTTP	Hyper Text Transfer Protocol
IAD	Integrated Access Device
IADMS	IAD Management Server
ICMP	Internet Control Message Protocol
IETF	Internet Engineering Task Force
ITU-T	International Telecommunication Union - Telecommunication
MDU	Multi-Dwellings Units
MGCP	MEDIA GATEWAY CONTROL PROTOCOL
MIB	Management Information Base
MTU	Multi-Tenants Units
NAT	Network Address Translation
NCS	Network-based Call Signal protocol
NGN	Next Generation Network
POTS	Plain Old Telephone System
PPPoE	Point-to-Point Protocol over Ethernet
PSTN	Public Switched Telephone Network
RARP	Reverse Address Resolution Protocol
RFC	Request for Comments
RTP	Real Time Protocol

SDP	Session Description Protocol
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol
TCP	Transmission Control Protocol
TELNET	Telecommunication Network Protocol
TFTP	Trivial File Transfer Protocol
UA	User Agent
UI	User Interface
UDP	User Datagram Protocol
URL	Uniform Resource Locator
VAD	Voice Activity Detection
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network

Appendix C Code Determination Flow Chart